DIAGNOSTIC PERCUSSION : AN INVESTIGATION USING ELECTRONIC MEASUREMENT TECHNIQUES

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C O N T E N T S

A B S T R A C T

Medical percussion, because of its simplicity, is an extremely useful diagnostic aid. Yet it is poorly understood. Confusion exists both in the terminology and in the description of the various percussion sounds. In addition, the sound pressure waveforms presently in use are found to be unsatisfactory.

A special purpose tape recording system, which was constructed to allow automatic retrieval of the waveforms onto chart paper is described.

Percussion sounds are first examined qualitatively. Preliminary observational studies of the waveshapes reveal the types and ranges of the sounds. No important features of these waveshapes are found to be obscured by quiet room noise.

Subjective descriptions of the sounds are reviewed. Results of Fourier analysis are correlated with judgements of frequency content. Other descriptions - intensity, duration, tactile sensation - are examined while producing a physical explanation of the sound pressure waveshape. Two fundamental physical components are isolated.

A four parameter model for quantitatively describing percussion sounds is proposed. The validity of the model is verified visually and acoustically with simulated waveforms; an electronic simulator being designed for this purpose. A sound reproduction system, developed to enable the transient percussion sounds to be reproduced, is discussed.

Computer-automated parameter measurement is used. Tape-recorded sounds are fed directly to the computer and the four parameters of each sound analysed are tabulated on the computer's teletype without any human intervention other than the initial setting up of equipment.

It is shown that 'resonant' sounds of varying quality can be satisfactorily quantified and that differences between sounds can be measured.

Variation in parameters due to changing the lung volume and to moving the percussion position down the chest are measured. Also, sounds from several healthy subjects are compared.

Success in detecting diseased tissue in the lung is achieved using this four parameter measurement technique.

CHAPTER 1

PERCUSSION AS A MEDICAL DIAGNOSTIC AID

1.1 INTRODUCTION

Today, at a time when methods of investigation in science and medicine are becoming more and more sophisticated, we often need reminded that there still exist many simple investigation techniques. Clinical percussion falls within this general category since it provides a quick and easy means of assessing the condition of the organs within the body.

Percussion, as the name implies, is an investigative method which involves tapping the body. This tap generates a percussion sound from which the medical practitioner is able to assess the condition of the organ lying below the percussion area. In the past, many different percussion techniques have been employed, but now the finger-finger method is almost universal. This involves placing the middle finger of the left hand (pleximeter) on the patient's chest and then striking the second phalanx of this finger with the middle finger of the right hand (plexor). The pleximeter finger firms the tissues in preparation for the impulsing blow which physicians say should be firm, but quick, with a rapid withdrawal; the entire effect coming from a wrist movement.

Another method of investigation is radiography, and if the ease and accuracy with which the results can be evaluated are considered, it is far superior to percussion. However, radiography requires bulky and expensive equipment, and specialist staff to operate it. Before any such sophisticated instrumentation is used, the medical practitioner will already have examined the patient and it is in this initial investigation that percussion is of immense value. Percussion is simple, requires no tools and no expense, and can be used anywhere. Experience on the part of the examiner is the only necessary requirement. If after this initial examination, the physician requires either confirmation or additional information, radiography will then be used.

Even although percussion is not as accurate as radiography, lesions in the lungs up to 5 cm deep and 2 or 3 cm in diameter can be detected (Delp 1968), and Roberts (1966) claims that during the evolutionary stages of chest disorders, percussion can reveal more than radiography.

1.2 HISTORICAL DEVELOPMENT

When percussion was discovered, it was a major advance in medical diagnosis, since at that time examination of a patient with internal disease was almost impossible. Previously, the only aids a physician had at his disposal were inspection and palpation, and so the examination of organs within the thoracic cage was automatically excluded. Therefore percussion for the first time enabled the condition of these organs to be determined while the patient was still living.

Percussion was the invention of Leopold Auenbrugger (1761), who, after seven years observation and experimentation at the Spanish Military Hospital in Vienna, published his findings in 'Inventum Novum'. The impact of his book however, even although it went through two editions, was never very great until Corvisart (1808) translated it into French and added his own commentary. Then followed an English translation by Forbes (1824) and a German one by Ungar (1845).

Auenbrugger's method was that of striking the patient's chest directly with the points of the fingers held close together. The patient himself firmed the surface skin and muscle by holding his head erect and throwing his shoulders back.

Sixty years later, this method gave way to mediate percussion when Piorry (1828) introduced the pleximeter. This was a hard striking board with which the physician firmed the surface tissues and so alleviated the need for any bodily contortion by the patient. Piorry's special pleximeter was an ivory disc about one and a half to two inches in diameter and at that time it proved to be the most popular type even although some other investigators made use of different materials for their pleximeters.

Later still, the pleximeter finger was introduced and gradually superseded its ivory counterpart. This finger was simpler to handle, and unlike the ivory pleximeter, just could not be lost or forgotten. These benefits were supplemented by a more important and yet not so obvious advantage. The ivory disc, being thinner and of greater surface hardness than the finger, tended to resonate on impact and so added colouration to the actual percussion sound.

Either one or two fingers could be used in striking the disc, but with the finger pleximeter, only one finger is used. This striking finger is known as the plexor. Special plexor hammers have also been employed and although for a time they proved quite popular, the finger has almost totally replaced them again.

The change over from ivory to finger pleximeter was taking place while Skoda (1839) was making his notable investigation into percussion. He was the first worker to make a serious attempt at giving percussion some physical basis. He examined the patient with extreme care and if the illness proved to be fatal he would check his diagnosis by a subsequent autopsy. From his inspection of the condition and structure of the organs he proceeded to reconcile his observations with the known laws of sound, showing that the 'fundamental quality' of the sounds depended on the presence or lack of air within the organs below the area percussed.

Percussion, by this time, had become normal practice in the examination of a patient. This can be seen from a number of books on physical diagnosis published at that time which contained extensive sections on percussion; Flint's (1856) perhaps being one of the better known English language versions.

Very few significant changes have since taken place in percussion technique, though from time to time there has been controversy over details. One such point concerned the magnitude of the force required to firm the tissues below the pleximeter. Cabot (1906) advocated a large force, so great in fact that a patient sitting without any back support required considerable exertion to avoid losing his balance. Recently, however, Casteleijn (1961) has shown that all that is really required is a force just great enough to firm the tissues and no more, by demonstrating that there is very little alteration of the sound intensity or pressure waveform once this critical force has been exceeded.

1.3 PERCUSSION SOUNDS

1.3.1 Classification of sounds

Initially, only two different types of sounds were distinguished -

Table 1.1 CLASSIFICATION OF PERCUSSION SOUNDS

SOURCE		Normal Lung 'Resonance'	Liver or Heart underlying Lung Decreased 'Resonance'	Below Lung Margin No 'Resonance'
Auenbrugger	1761	Sonus Clarior	Sonus Obturior	
Forbes	1824	Sonorous	Morbid	
Laennec	1826	Clear		Dull
Flint	1876	Resonant	Dull	Flat
Cabot	1906	Vesicular Resonance	Dull	Flat
Major	1956	Resonant	Dull	Flat
Appleton	1958	Resonant	Less Resonant	Dull
Lewis	1966	Resonant	Impaired	
Macleod	1967	Resonant	Impaired	Dull
Delp	1968	Resonant	Dull	Flat

'sonorous' and 'morbid' (Forbes 1824). As this did not prove altogether satisfactory, other categories had to be added. Piorry (1828) introduced terms such as 'liver sound', 'kidney sound' and even 'spleen sound', all of which he claimed to be able to detect. Skoda (1939), however, showed this to be an erroneous and fanciful idea and so introduced his own classification. He distinguished four 'qualities' of each sound and his groupings were the various combinations of these four 'qualities'. Even although Chapman (1943) still found this classification valid, it never proved to have any practical usefulness due to the difficulty of defining the four 'qualities', let alone measure them. Hence, rather than describe all percussion sounds with a number of characteristics it became accepted practice to identify certain types of sound; each type containing all sounds which were sufficiently similar in quality to be described under one term. Those terms which have come into common usage are (in decreasing 'quality of resonance') 'tympanic', 'hyperresonant', 'resonant', 'impaired resonance', 'dull', 'flat' and 'stony dull'.

'Resonant' refers to the quality of the percussion sound over the air filled lung, under normal healthy conditions of the chest. When the heart or liver underlies the normal lung, a decrease in 'resonance' is noted and below the lung border the 'quality of resonance' disappears.

At first sight, the categories of sounds seem quite adequate but since the sounds have not yet been described properly, many physicians use the same term for different sounds. Table 1.1 illustrates this point over the various uses of the term 'dull', sometimes describing 'decreased resonance' and sometimes no 'resonance'.

1.3.2 Description of sounds

The description of the various sounds listed above, at the best of times is quite vague and generally the sounds are described with reference to the 'resonant' sound which is said to be a 'low pitched sound'. 'Increased resonance' is said to result in an even lower pitched sound, and 'decreased resonance' in a shorter, quieter sound with 'high frequency vibrations'. A sound containing no 'resonance' has minimal duration and intensity, and again is described as a 'high frequency vibration'. One observer (McKusick 1955) however did make the observation that it contained mainly the sound of impact.

To avoid further confusion between the various sounds, the qualitative term 'dull' will refer, in this thesis, only to those sounds exhibiting no 'resonant quality', as obtained where there is no underlying air cavity.

The 'tympanic' sound has its own individual characteristic, in that it contains a more distinctive single frequency component than the others.

One of the main points which must be emphasised about the description of the sounds is that there is no absolute description of any sound, they are all described relative to other sounds. The descriptions are also couched in such vague terms that often textbooks on physical diagnosis, such as that by Macleod (1967), only tabulate the various sounds without making any attempt whatsoever at describing the sounds.

1.4 USE IN DIAGNOSIS

So far, the various sounds have only been considered with reference to normal sounds, but the effects of disease must now be considered since it is the changes in sound resulting from disease which make percussion useful in medical diagnosis. Innumerable diseases change the percussion sounds from the 'normal' but only a few will be mentioned here as illustrative of the conditions under which certain types of sound can be expected.

'Resonance' over the normal lung increases to 'hyperresonance' when a greater volume of air than normal is contained under the area percussed, such as over emphysema, or over the normal lung when the other lung is consolidated or over the normal part of a lung in lobar pneumonia. Decreased 'resonance', and in some cases 'dullness', results from pneumonia, fibrosis or tuberculous consolidation of the lung and also from a pleural effusion or fibrosis.

When the physician percusses the chest he continually compares one part of the chest with another, especially one lung with the other, and it is the difference in 'quality' of the sound that he recognises rather than any absolute 'quality'.

The change in position of the lower lung border is also indicative of certain disorders. In emphysema, the border is lower than normal and with diaphragmatic pleurisy or enlarged liver, the border is higher.

1.5 ANALYSIS OF PERCUSSION SOUNDS

1.5.1 Recordings

A major step in any analysis must necessarily be that of obtaining a permanent record of the sounds, and hence allow the various sounds to be studied in detail.

Recording the percussion sound pressure waveform was first attempted by Castex (1895), who modulated a manometer flame with the percussion sounds. Gas was fed into one tube, through a manometric capsule and burned at the end of a second tube, enabling pressure waves at the capsule to modulate the flame intensity and height. This capsule was placed in contact with the trachea or inside the mouth of the patient whose chest would then be percussed. Experimenting with corpses, two trocars (triangular tipped instruments used for removing fluid from cavities) were forced through the chest wall, penetrating the pleural cavity which then functioned as a capsule. Recordings were made on photographic plates, mounted on a moving carriage and pulled by hand in front of the flame.

Another method was devised by Selling (1907) in which he first recorded the sounds onto a gramophone disc. Then on playback, the vibrations were transmitted through a system of compound levers and hence marked onto the smoked paper of a kymograph.

Selling also employed a microphone and an Einthoven string galvanometer to obtain recordings. Electrical signals from the microphone forced the rotation of a silver quartz filament, suspended between the poles of a strong electromagnet. Light from an arc lamp was reflected from the filament and recorded on a moving photographic film.

Despite the progress which has been made in electronic sound recording, very little use has been made of the application of this technique to percussion. Indeed, a textbook on physical diagnosis by Delp, published as recently as 1968, still includes Selling's original recordings.

1.5.2 Analysis

To the first investigators, the sense of pitch of the percussion sounds was considered to be their most important feature, so they tried to determine the 'fundamental frequency' of the sounds using their own natural sense of pitch. Later, to obtain a more 'accurate' measurement, the sounds were compared with frequency standards; the most common being Helmholtz resonators and tuning forks. Gerhardt (1876) was one of the first workers to use Helmholtz resonators, and many years later this method was still being used by Selling (1907) and Martini (1922). The concept of 'fundamental pitch' is quite inadequate and does not describe pitch in the true sense of a particular frequency being present, but rather, the subjective impression that the percussion sound gives the hearer. This is discussed further in section 4.4.

Attempts were also made at determining the intensity of the different sounds. Selling, when comparing 'resonant' and 'dull' sounds, compared the maximum distances at which these sounds were just audible and with the help of a microscope, compared the depth of the groove these sounds cut when recorded on a phonograph cylinder.

The duration of the sounds has also been studied. Selling, from his galvanometer recordings estimated durations of 420 ms for 'resonance' and 280 ms for 'dullness'. In the light of the investigation of this present study (see section 4.6) where durations of the order of 15 ms for 'resonant' sounds were found, the figures quoted above are quite surprising and can only raise doubts as to the recorded waveforms actually being a faithful reproduction of the input sound pressure waveforms, and not the response of the recording system to the given input. The output of any recording system is necessarily a characteristic of that system as well as of the input signal. Hence it is necessary to employ a system which produces an insignificant change on the sound pressure waveforms. This is achieved by having a smooth frequency response in the passband of the input signals and by introducing negligible noise compared to the sound noise already present with the percussion sounds. Great care was taken along these lines in producing a suitable recording system for the present investigation.

Some time after Selling had ended his work investigators returned to analysing the sounds in the frequency domain with an attempt this time to produce a complete frequency spectrum of the sounds rather than just a 'fundamental frequency'. McKusick (1955) made use of sound spectroscopy by first recording the sounds on the rim of a magnetisable disc and then playing them back through a variable frequency filter. Each rotation of the disc stepped the centre frequency of the filter, whose output was recorded as a function of time on electrosensitive paper rotating in synchronism with the disc; the marking position also being stepped. The final result was a two dimensional sound spectrograph with time and frequency axés; the depth of shading being proportional to the magnitude of any frequency component at a particular instant in time.

Casteleijn (1961) extended this method to obtain a Fourier spectrum of the different sounds. Using his technique he analysed various sounds from the chest and stomach. Unfortunately, he found that the difference between the frequency spectra of 'resonant' and 'dull' sounds were not clearly observable.

1.6 TACTILE SENSATION

So far, only sound has been discussed, but the sensation felt by the fingers during percussion is also of significance to many physicians.

Table 1.2 DESCRIPTION OF TACTILE SENSATION

SOURCE		DESCRIPTION
Laennec	1826	Sense of elasticity (direct percussion)
Piorry	1828	Sense of resistance
Skoda	1839	Sense of resistance
Cabot	1906	Sense of resistance
Coleman	1939	Vibration sense
Major	1956	Ability to perceive differences in frequency of waves set in motion by percussion
Appleton	1958	Underlying resistance/range of vibrations
Casteleijn	1961	Vibratory sensation

In fact, this tactile sensation is held by some to be of greater importance than the sound heard.

Although this tactile sensation had been noticed before the advent of mediate percussion (Laennec 1826) it was not until Piorry stressed its importance that notice was taken of it.

Originally, tactile sensation had been described as the 'sense of resistance' felt on percussion (Table 1.2), with Cabot (1906) equating 'resonance' with an 'elastic resistance' and 'dullness' with a 'woodeny feeling'. Nowadays, the use of frequency analysis has so gripped investigators that the term 'vibratory sensation' has arisen with the idea that the pleximeter, through its sense of touch, detects the particular frequency of vibration produced on percussion; low frequencies with 'resonant'ssounds and high frequencies with 'dull' sounds. Section 4.5.5 of the present investigation recommends that the term 'sense of resistance' be returned to.

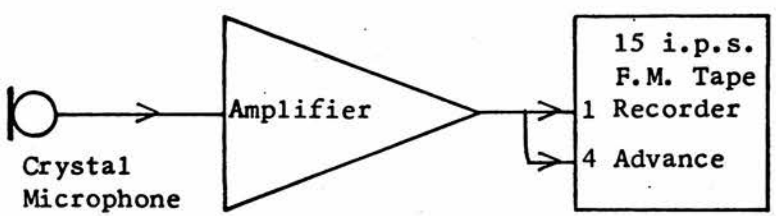
1.7 AIM OF RESEARCH

The aim of this research project has been first to make a qualitative study of the percussion sounds, then to discover how far these sounds could be described in quantitative terms in such a way as to make a measurable distinction between the various sounds hitherto classified in vague qualitiative terms.

Primarily, this research project has dealt with percussion sounds, with reference being made to tactile sensation only when this arose out of the analysis of the sounds.

Figure 2.1 COMPLETE SYSTEM

Record



Playback

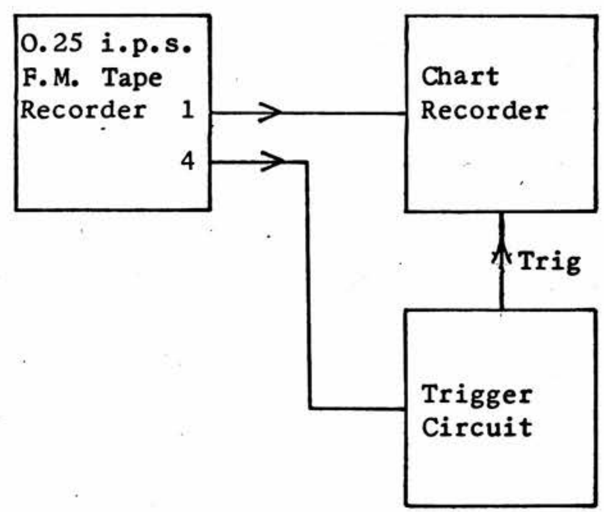
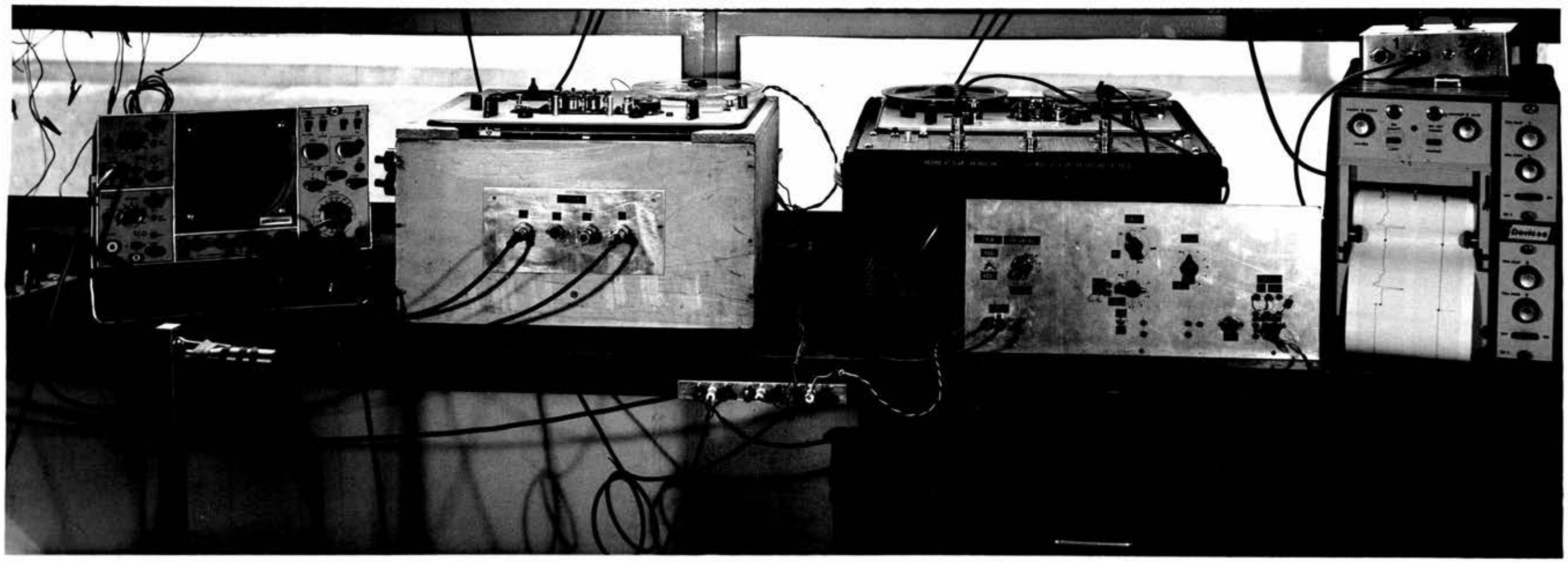


Figure 2.2 COMPLETE RECORDING SYSTEM



CHAPTER 2

S Y S T E M   F O R R E C O R D I N G   P E R C U S S I O N   S O U N D S

2.1 COMPLETE SYSTEM

Shown in block diagrammatic form in figure 2.1 is the complete system which was developed for recording percussion sounds. A photograph of the equipment appears in figure 2.2.

Percussion sounds which had been generated on the human body travelled outwards to be picked up by the microphone. The pressure variations of the sounds were then converted by the microphone into an equivalent electrical waveform. Before being recorded on magnetic tape, the waveforms were amplified to some suitable voltage level. On later playback of the tape, the percussion sound waveforms were recorded in a permanent visual form on chart paper. The trigger circuit linking a second channel of the tape recorder and the chart recorder was used to start the chart paper moving automatically, before a waveform was to be produced and then to switch it off again after some preset time, which was sufficient to record all or part of the waveform as desirable.

The oscilloscope shown in figure 2.2 was used in the initial setting up of the voltage levels and for monitoring the waveforms being recorded to ensure that acceptable voltages were being presented to the tape recorder.

Considerable care was taken over the design and development of the various sections of the recording system to ensure that the sound pressure waveform was not being distorted either within an individual section or by an improper matching of two sections. Of most importance in this respect was a consideration of frequency response and noise levels within the system. Care was taken to ensure that the frequency response of the system was able to pass the various percussion sounds with negligible distortion. Concerning the noise, the final design produced substantially less than that resulting from typical quiet room background noise.

2.2 MICROPHONE

After a consideration of both cost and performance of various microphones, a high quality piezoelectric microphone, Brüel and Kjaer type 4117, was purchased. Like other microphones of this type, the output voltage produced across the ceramic crystal was directly proportional to the mechanical strain set up in the crystal and hence also to the sound pressure.

The performance of the microphone had to be such that the voltage waveform faithfully reproduced the sound pressure waveform received; it being important not to discard any information which the waveshape might contain. In any case, it is generally the best policy to ensure that the output waveform be independent of the measuring system, particularly where a study of the waveform is desirable. This necessitated the use of a microphone with a flat bandpass response within the limits of the various percussion sound frequency components. The free field response of the microphone under consideration (calibrated before being despatched by the manufacturer) varied less than $\pm$ 0.5 dB from 10 kHz down to its lower calibration frequency of 10 Hz. The actual low frequency cut-off extended below 10 Hz and depended on the microphone amplifier as well as the microphone.

With a view to determining the maximum significant high frequency component of the various waveshapes, the microphone was connected both directly and through a low pass RC filter to separate channels of a storage oscilloscope and the effect on the percussion waveform of filtering at various frequencies observed. On close visual comparison of the direct and filtered waveforms for both the overall waveshape and for small detail, no visible difference could be detected on the oscilloscope screen with the filter cut-off set at 1.5 kHz. The suitability of 1.5 kHz for the minimum high frequency cut-off of the system was further substantiated when Fourier analysis (section 4.3) confirmed that the frequency components of various percussion sounds at 1.5 kHz lay well below the maximum frequency component of each sound. The following are typical figures

Percussion Sound	Relative magnitude of 1.5 kHz frequency component compared with maximum frequency component
'resonant'	- 55 dB
'tympanic'	- 50 dB
'dull'	- 32 dB

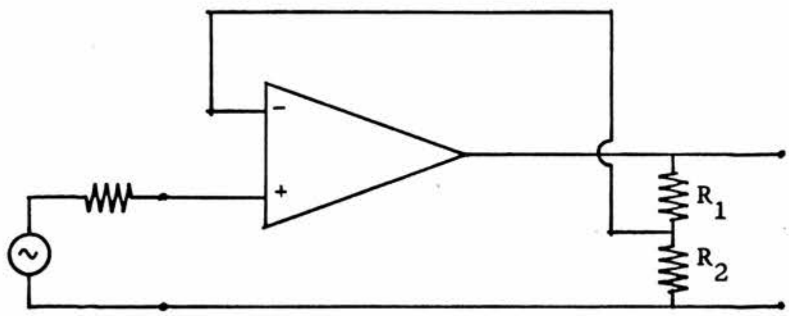
Hence the 10 kHz high frequency response of the microphone was more than adequate for the proposed work.

At the low end of the frequency scale the microphone had again to be considered to decide if its low frequency response was adequate. Two features determine the lower cut-off frequency of a crystal microphone; one is the construction of the microphone itself and the other is due to the combined effect of microphone capacitance and input impedance of the amplifier to which the microphone is connected. This latter effect will be considered in the following section. On the microphone, a capillary tube of 0.25 mm internal diameter allowed for air leakage and so provided for equalisation of static pressure on both sides of the diaphragm at a predetermined rate; the time constant of pressure equalisation was 0.05 s corresponding to a 3 dB cut-off of 3 Hz. Even although 3 Hz was well below the lower limit of hearing, the frequency cut-off of the system had to be chosen with care, so as to preserve the original pressure waveform. Distortion produced by an inadequate low frequency response was easily observed as changes in waveshape. This was because the waveform tended to be differentiated on the rising gradient of the frequency transfer function.

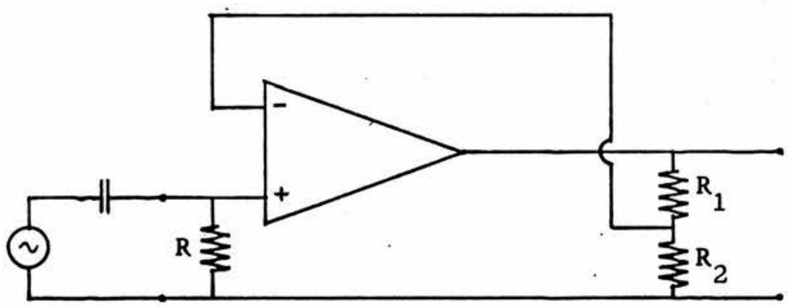
For the purpose of determining the required low frequency cut-off, the low frequency response of the microphone was extended below 3 Hz by inserting a wire in its capillary tube. This had the effect of reducing the rate at which the pressures on both sides of the diaphragm were equalised and hence also lowered the cut-off frequency. With the microphone connected to a suitable amplifier the output waveforms before and after passing through a high pass filter (6th order Butterworth) were compared. The comparison was made by visual inspection and by the measurement of the heights of prominent peaks on the waveform, which became easily distorted if the lower cut-off frequency was too high. With the filter cut-off at 3 Hz the change in the relative heights of the peaks was barely measurable on the chart recordings produced, and in any case less than 3%. At 5 Hz, the change in the relative heights

Figure 2.3 NON INVERTING AMPLIFIER USING OPERATIONAL AMPLIFIER

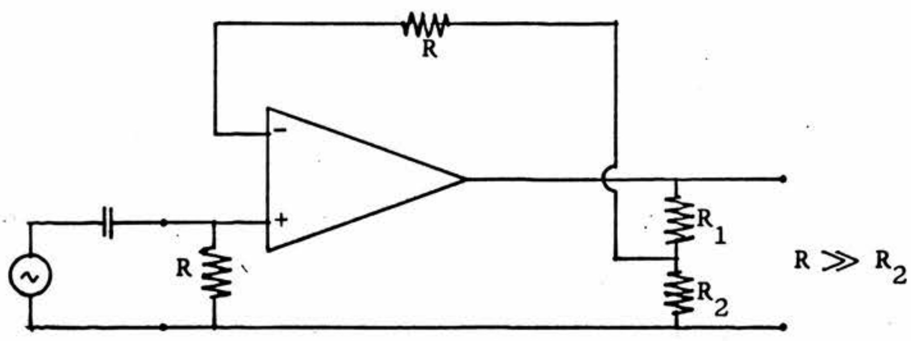
(a) Voltage Source



(b) Piezoelectric Microphone Source



(c) Design for Temperature Stability



did just become observable and at 10 Hz was typically as much as 25%.

For practical reasons the low frequency cut-off had to be as high as possible to avoid the low frequency pressure waves generated by, for example, wind or the closing of doors; both of which produce extremely large pressure changes and easily generate voltages outwith the acceptable limits of the recording system. Therefore the microphone's response was returned to its original low frequency cut-off of 3 Hz, which was then retained as the low frequency 3 dB point of the total system.

2.3 AMPLIFIER

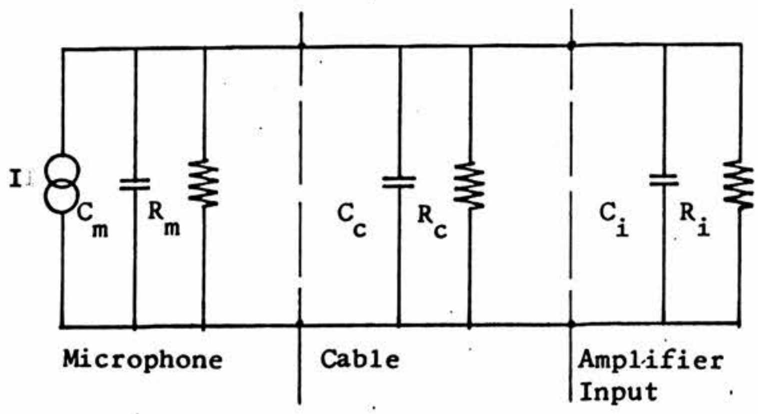
The electrical waveform had next to be amplified from the few millivolts peak output from the microphone, of a typical percussion sound, to a voltage more commonly accepted by other electronic apparatus. For example, the tape recorder described in the following section accepted signals within the $\pm$ 1 V range, and the greater the use made of this range, the better the signal-to-noise ratio obtained.

2.3.1 Operational Amplifier Method

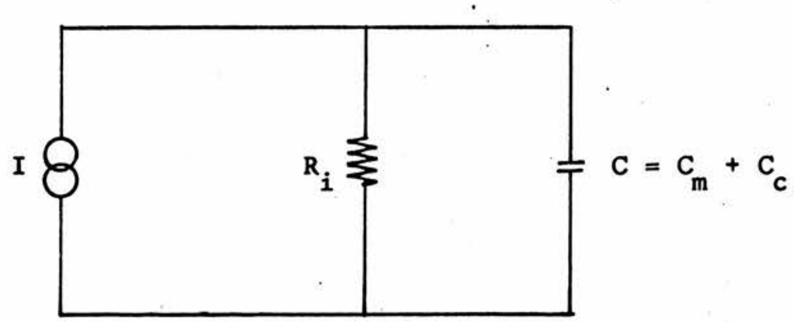
The microphone produced a positive voltage output for an increase in pressure, and this being a useful convention to retain the amplifier had to be of the non-inverting type. Modern integrated circuit operational amplifiers allow this type of circuit to be constructed relatively easily. Referring to the diagram of figure 2.3(a) it can be seen that the feedback resistors constrain the output voltage to be $\left(\frac{R_1 + R_2}{R_2}\right)$ times the voltage on the non-inverting terminal. Also, due to the very high gain of operational amplifiers only a very small differential voltage appears across the inputs for an output voltage within its range of operation. Hence the gain of the amplifier will

Figure 2.4 EQUIVALENT CIRCUIT OF MICROPHONE, CABLE AND AMPLIFIER

(a) Complete Equivalent



(b) Simplified Equivalent



closely approximate to $(\frac{R_1 + R_2}{R_2})$.

In figure 2.3(a) the input bias currents are provided for, through the feedback network and through the source resistance, but if the microphone replaces the voltage source a current path must be retained by adding a resistance as shown in figure 2.3(b).

2.2.2 High input Impedance

The equivalent microphone/amplifier input circuit is drawn in figure 2.4(a); the microphone being shown as a current generator. The symbols used are as follows:-

C_m = capacitance of microphone = 3800 pF

C_c = capacitance of cable = 240 pF

C_i = input capacitance of amplifier

R_m = leakage resistance of microphone

R_c = leakage resistance of cable

R_i = input resistance of amplifier

Since R_m and R_c were very much greater than the input impedance of the amplifier and since care was to be taken to keep C_i to a minimum, they were neglected in the initial calculations. (After construction of the amplifier C_i was found to be only 40 pF). The resulting simplified circuit is shown in figure 2.4(b) where $C = C_m + C_c$

$$= 4.0 \text{ nF}$$

To preserve the microphone response the microphone input circuit time constant had to be greater than 0.06 s. Using an amplifier input resistance of 22 MΩ gives a satisfactory time constant of 0.09 s.

To obtain an input impedance of 22 MΩ, that of the operational amplifier itself had to be greater than 22 MΩ. FET input operational amplifiers adequately fulfil this requirement and the particular one selected (integrated circuit type 144A) had a differential and common mode resistance of $10^{11}\Omega$.

In selecting the operational amplifier, temperature drift instability had to be considered. This drift is caused by a changing input bias current developing a changing voltage across the input resistance. The 144A specifications quoted a maximum bias current of 100 pA at 25°C, doubling every + 10°C. Since the voltage drift at the input is directly proportional to the input resistance, it was advisable to make R (see figure 2.3) no higher than necessary, which in this case was 22 MΩ, and hence the expected offset voltage at 25°C was 2.2 mV. At high amplifier gains this voltage, although small, could easily have introduced instability problems. Fortunately, to a certain extent this can be overcome since both input currents are approximately equal and change in similar fashion with temperature, and so if each sees the same resistance (figure 2.3(c)) the input bias voltages will tend to follow each other, resulting in a very small differential input voltage.

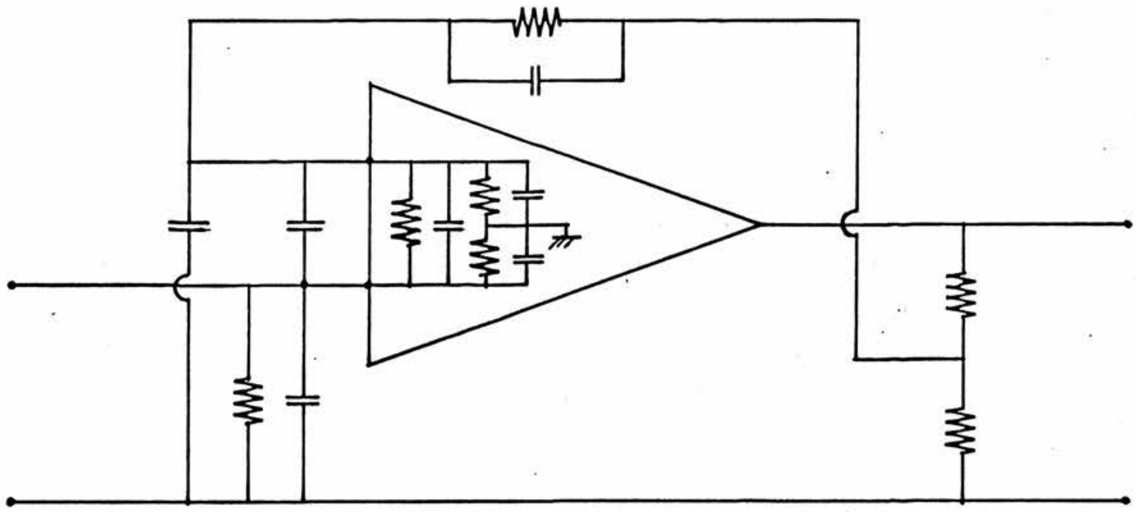
With its 60 dB gain, the final amplifier had a satisfactory temperature drift characteristic. After a warming up period of thirty minutes, during which the output voltage drift could reach 200 mV, the drift settled to a satisfactory value of 50 mV maximum peak-to-peak swing in any hour with typical ambient temperature fluctuations. This corresponded to a change in differential input voltage of only 50 μV.

2.2.3 Stray Capacitance

Initially the amplifier, built from the circuit of figure 2.3(c), did not yield a smooth frequency response in its pass band as had been expected. This deviation from the original theoretical analysis

Figure 2.5 EFFECT OF STRAY CAPACITANCE

(a) Complete Circuit



(b) Simplified Circuit

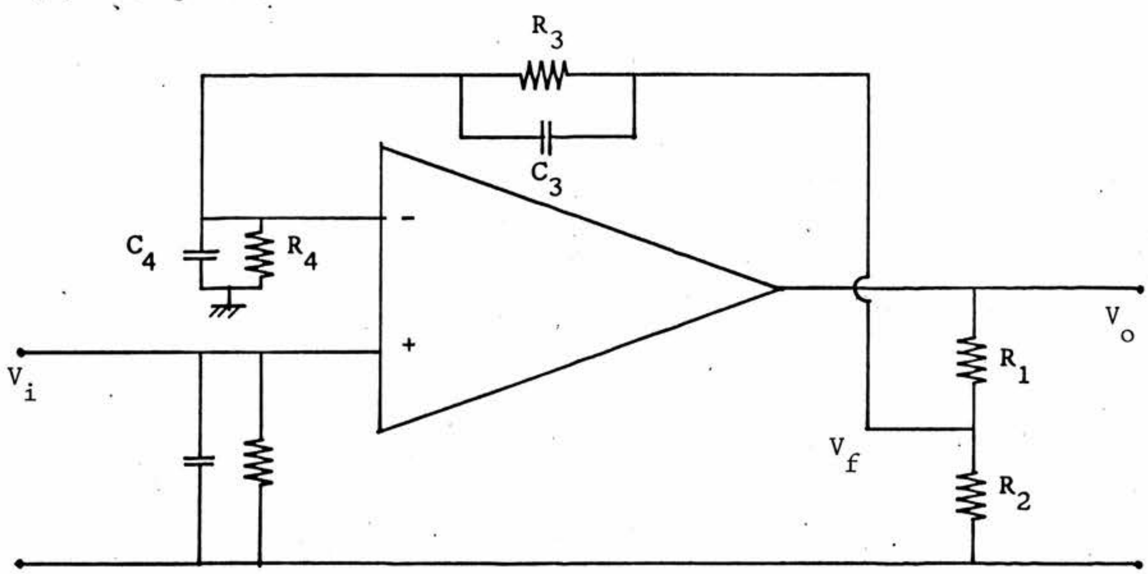
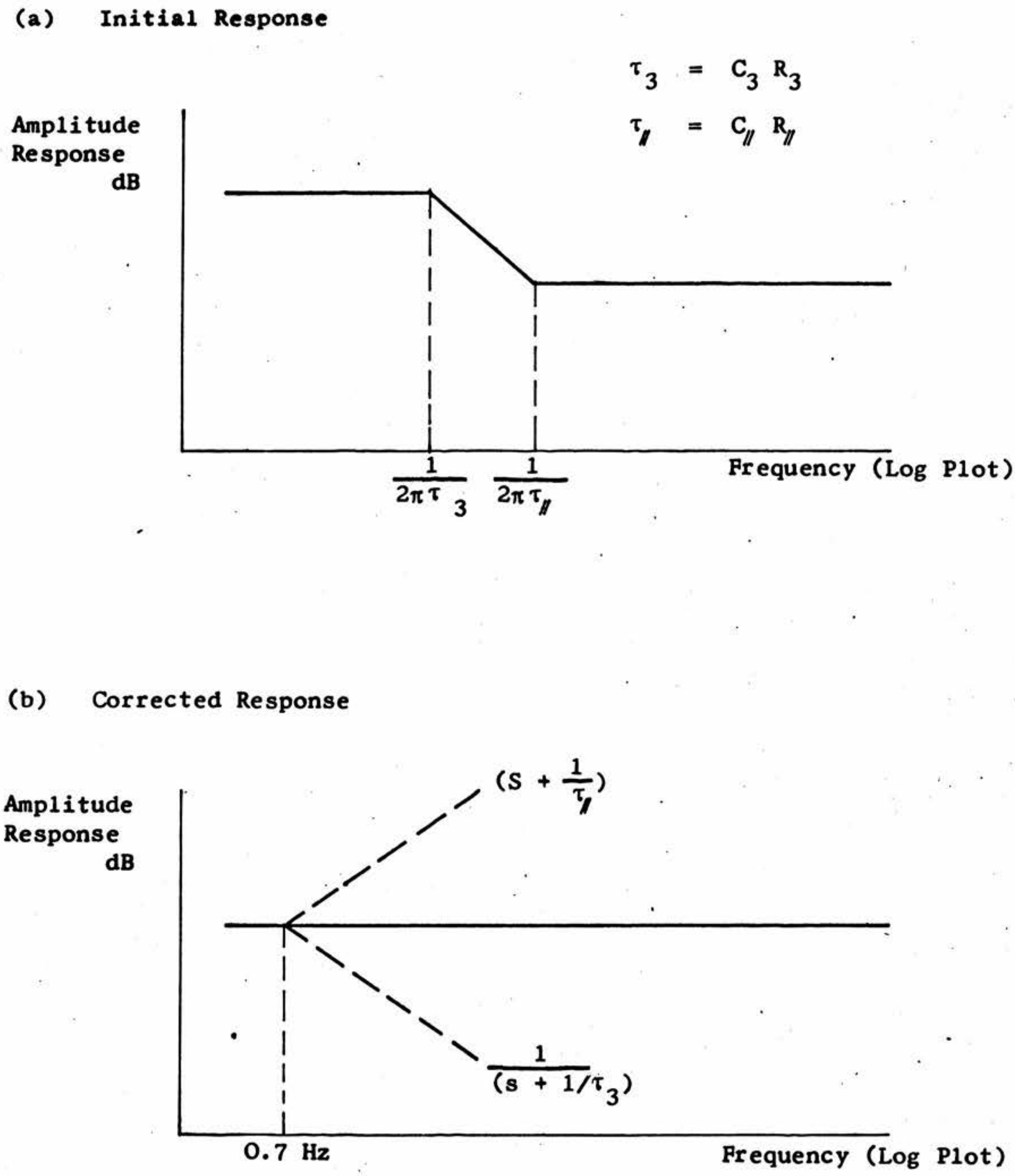


Figure 2.6 AMPLIFIER BODE PLOTS



was due to the effect of stray capacitances appearing across the high resistance components. Figure 2.5(a) is the result of redrawing the circuit with the stray capacitances added across all high resistances. Before attempting an analysis, the circuit was first of all simplified to that shown in figure 2.5(b).

Now, since the feedback resistors were very much smaller than R_3 or R_4, the voltage at the junction of these feedback resistors was taken to be independent of R_3, C_3, R_4 and C_4. Therefore the transfer function was found from

$$\frac{V_f}{V_i}(s) = \frac{C_3 + C_4}{C_3} \quad \frac{s + 1/(C_{/\!/} R_{/\!/})}{s + 1/(C_3 R_3)}$$

where $\quad C_{/\!/} = C_3 + C_4$

$$R_{/\!/} = \frac{R_3 R_4}{R_3 + R_4}$$

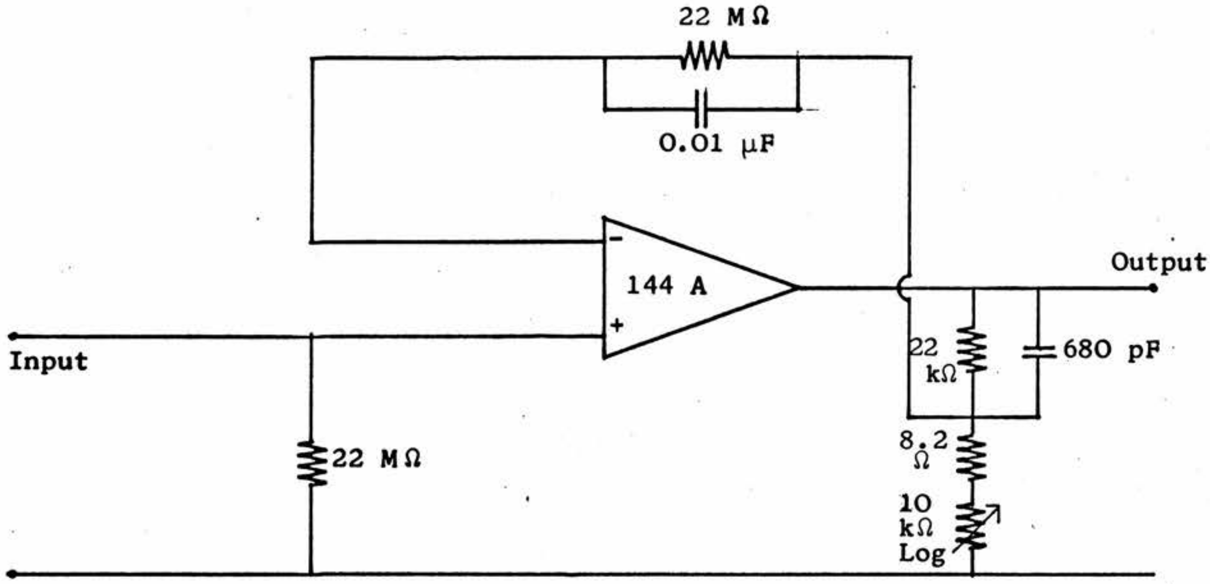
and hence

$$\frac{V_o}{V_i}(s) = \frac{C_3 + C_4}{C_3} \cdot \frac{R_1 + R_2}{R_2} \quad \frac{s + 1/(C_{/\!/} R_{/\!/})}{s + 1/(C_3 R_3)}$$

The frequency response resulting from this type of transfer function is shown as a Bode magnitude plot in figure 2.6(a). Bode plots use logarithmic scales which simplifies their construction and interpretation; hence the reason for the x-axis of the figure being scaled in decibels. A Bode plot is a plot of the asymptotic magnitude-frequency response of the system, with the break points occuring at the frequencies where the imaginary part equals the real part within the various products of the transfer function.

On the Bode plot shown, the first break point was caused by C_3 and

Figure 2.7 FINAL AMPLIFIER DESIGN



R_3. Above the break point frequency$\left(\frac{1}{2\pi\ C_3\ R_3}\right)$ the plot took on a - 6dB octave slope because of the effect of the pole resulting from $\left(s + \frac{1}{C_3\ R_3}\right)$ in the denominator of the transfer function. The next break point was due to the effect of the capacitors C_3 and C_4 in parallel, and the resistors R_3 and R_4, also in parallel. Above this frequency the response levels out due to the effect of the zero caused by $\left(s + \frac{1}{C_{\parallel}\ R_{\parallel}}\right)$.

Both break points lay in the frequency range which was required to be flat. As it was not possible to substantially reduce the stray capacitance, even with further care over component layout, some other solution had to be found.

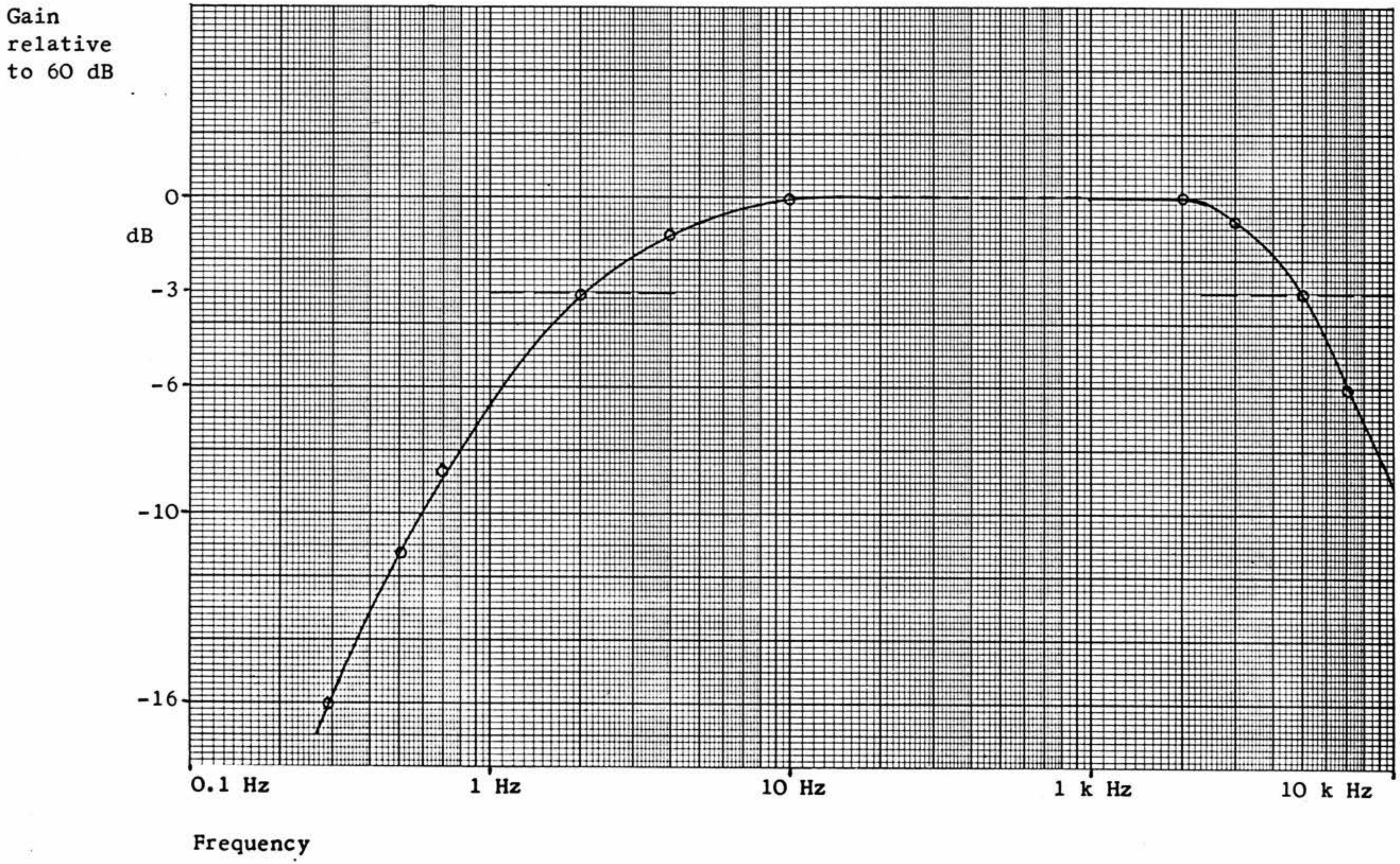
By adding a large capacitor (0.01 μF) across R_3, the value of C_3 was increased so much that the time constants $C_3\ R_3$ and $C_{\parallel}\ R_{\parallel}$ became approximately equal (0.22 s). This resulted in both break points converging on 0.7 Hz as shown in the Bode plot of figure 2.5(b), producing a flat frequency response as required.

2.2.4 Final Design

The final design of the amplifier is shown in figure 2.7. To prevent any tendency towards instability a 15 μs time constant was introduced by adding a 680 pF capacitor across R_1; this corresponded to a high frequency cut-off of 10 kHz. The actual limit of the high frequency 3 dB point, determined by the 60 dB gain of the amplifier, was 5 kHz and lay above that required (1.5 kHz).

Amplifier noise was the next important consideration. The equivalent input noise measured on an r.m.s. voltmeter covering the entire frequency range of the amplifier except that below 10 Hz, was found to be only 2.0 μV. It should be noted that Johnson noise of

Figure 2.8 AMPLIFIER RESPONSE



the two 22 MΩ resistors was shorted to ground through the 0.01 μF capacitor and the 4 nF equivalent input capacitance: if this had not been the case each resistor would have contributed 42 μV r.m.s. Lying considerably below the equivalent quiet room input noise for the given microphone, the noise performance of the amplifier was also satisfactory.

The microphone had already been tested by the supplier and was known to operate satisfactorily, and so the task of checking it did not need to be performed. Nevertheless, to test the response of the equivalent microphone-amplifier combination, a signal generator was connected to the amplifier through a capacitance equal to that of the microphone-input capacitance (4 nF) and the frequency response of figure 2.8 obtained, proving that the amplifier was working satisfactorily and that the lower frequency limit was determined only by the microphone itself, since the cut-off measured lay below the required 3 Hz.

2.4 TAPE RECORDER

2.4.1 Frequency Response

To prevent the necessity of producing percussion sounds from the body every time a particular study or analysis was to be carried out, it was considered important to be able to store the percussion sounds and so allow an indefinite number of replays of any sound or group of sounds.

Magnetic tape is an excellent medium for such storage since it allows the reproduction of sounds to be achieved with ease. Wastage of tape is prevented since tapes can be rerecorded if desired.

Several magnetic tape recording techniques are now widely used; the most popular being direct recording and frequency modulation recording.

Direct recording, although being simple and giving the best high frequency response for a given tape speed, yields a poor low frequency response. On the other hand frequency modulation techniques extend the frequency response down to d.c. As already discussed, it was of considerable importance that the low frequency response of the entire system should not tail off above 3 Hz, which immediately ruled out direct recording, and so frequency modulation, having no such undesirable defect, was selected.

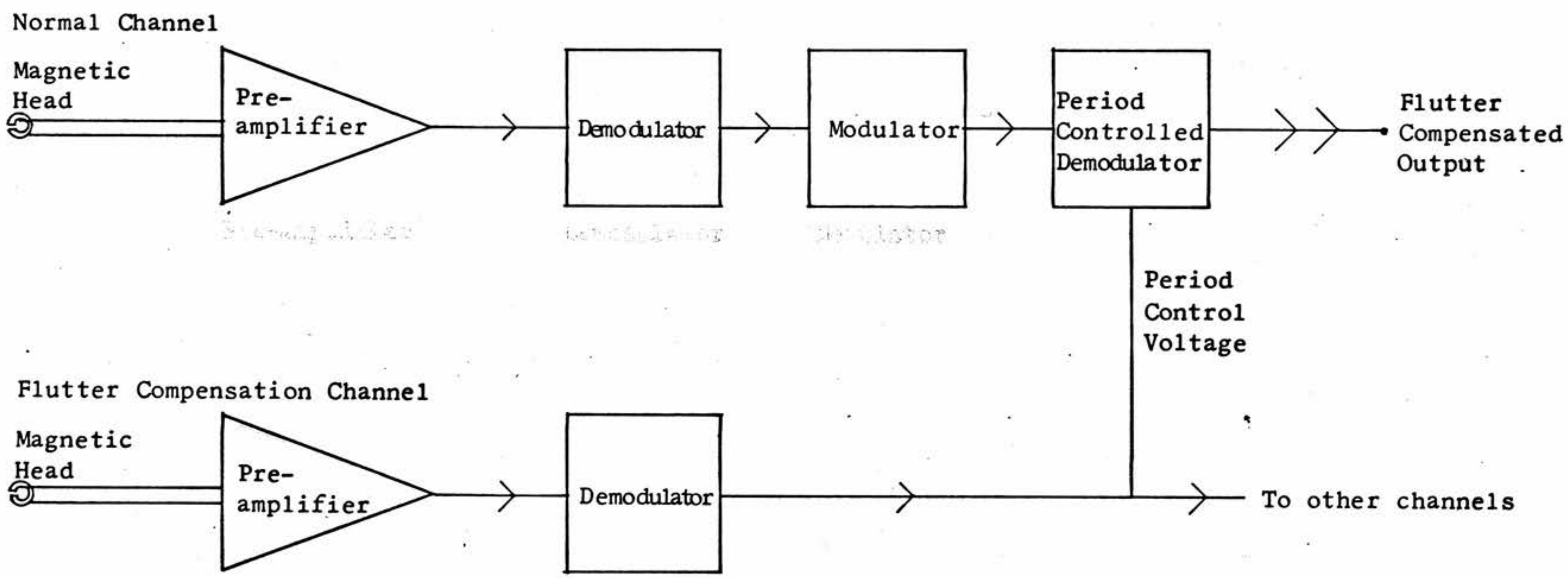
A tape recorder, running at 15 i.p.s. and have a frequency response from d.c. up to 2.5 kHz, was built to be compatible with a slow speed (0.25 i.p.s.) recorder already available.

Tape recording offers far more advantages than just information storage. For example, the facility to replay at different speeds greatly enhances its usefulness. In this particular work, slow replay was an invaluable aid when using the tape recorder in conjunction with a mechanical device such as a chart recorder with its inherent slow response. The chart recorders used (section 2.6) had an upper frequency limit of 80 Hz and so the frequency bandwidth of the input signal (1.5 kHz maximum) had to be reduced. Reducing the bandwidth sixty times results in an upper frequency of 25 Hz, ideal for chart recording. To obtain such a reduction requires the tape to be replayed at one sixtieth of the recording speed, that is 0.25 i.p.s.; a replay speed which was already available.

2.4.2. Flutter

Next to the frequency response, the noise generated by the tape recorder was considered to be of most importance. Generally the

Figure 2.9 BLOCK DIAGRAM WITH FLUTTER COMPENSATION



largest noise contribution is that of wow and flutter - wow referring to low frequency variations of a few Hertz and flutter to all higher frequency ones, although both are usually referred to collectively under the one name, flutter. Flutter arises from irregularities in tape speed, which become more pronounced at low speeds. Among the many factors contributing to flutter the following must be included; irregularities in the mechanical system such as eccentricities in the capstan or imperfections on the drive system, and the effect of non uniform friction acting on the tape.

In the system under discussion flutter was corrected for, by using one of the four channels (channel 3) as a flutter compensation channel. While recording, zero volts was applied to this channel, such that on playback the flutter compensation channel contained only the flutter voltage, while the other channels contained both the flutter voltage and the recorded signal. Hence the flutter channel was used to compensate for the flutter on the other three channels.

A block diagram of the playback system is shown in figure 2.9. The demodulated output of the flutter compensation channel was fed to the period controlled demodulator of each of the other three channels. At the centre of the demodulator was a monostable which was triggered at a frequency which was directly proportional to the output voltage of a channel prior to flutter compensation. The period of the monostable was controlled by the output voltage from the flutter compensation channel in such a way as to offset the effect which the flutter, superimposed on a normal channel, had on the recorded signal. Following the monostable was a low pass filter which gave an output proportional to the mark-to-space ratio (on-time to off-time) of the monostable, and hence resulted in a flutter compensated output.

Even at the slow replay speed of 0.25 i.p.s. a signal-to-noise ratio of 40 dB was obtained, and although the output noise voltage was greater than the amplifier noise with a gain of 60 dB it was still less than the equivalent output voltage of the system from room noise under quiet conditions.

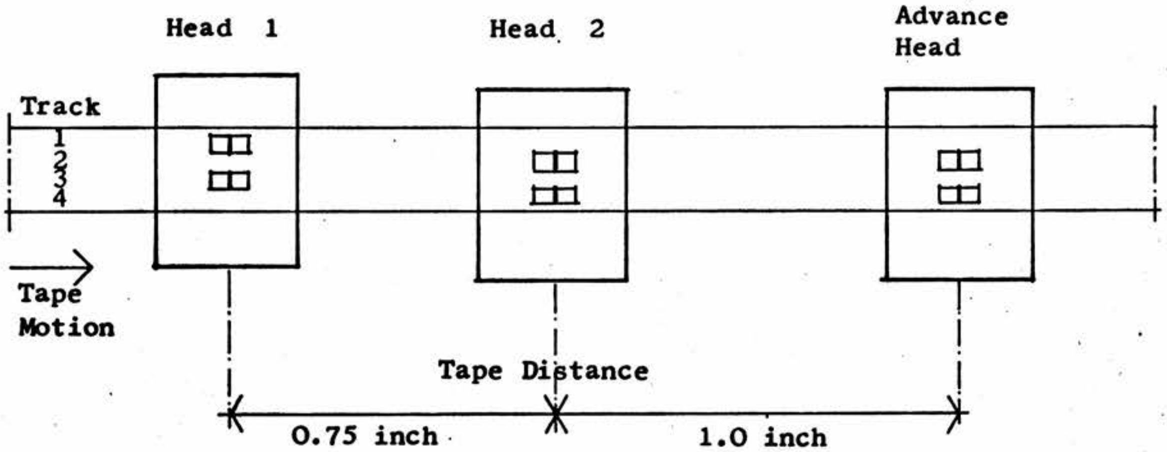
When measuring sound noise a physical measurement of sound pressure level can be taken, or a subjective measurement of loudness level can be made by using a standard frequency weighting network. Even with no frequency weighting in the sound pressure level measurement, frequencies below 20 Hz are often disregarded unless special consideration is being given to those lower frequencies.

In this present case a subjective loudness level was not required, as a measure of the effect the noise had on the recordings had to be found, and not the effect of the noise on the human observer. This meant that the noise contribution over the total frequency range of interest had to be considered; from 3 Hz to 1.5 kHz.

Using the averaging system (section 4.2) a print-out was obtained of five series of one hundred sequential samples of typical room noise under very quiet conditions; the interval between samples was set to allow all frequency components up to 2 kHz to be retained. After calculating the mean value of the standard deviations of all sequences of samples, this value was converted to an equivalent sound pressure level. Quiet room noise was found to add a much greater contribution to the system than that of the tape recorder. Hence the noise limitation was not that of the system but of the external room noise.

In the table below, the equivalent sound noise of various sources

Figure 2.10 RECORDING HEAD ARRANGEMENT



in the system are listed to enable a comparison to be made.

Source	Output Noise (r.m.s.)	Measurement Conditions	Equivalent Sound Noise (r.m.s.)
Amplifier	2 mV	60 dB gain	0.0007 Nm^{-2}
Tape Recorder	6 mV	60 dB amplifier gain	0.002 Nm^{-2}
Quiet Room Noise			0.009 Nm^{-2}

2.4.3 Recording Head Arrangement

Each recording head contained two magnetic gaps and hence could record on two channels. The first head was raised one track width above the second, such that the first head recorded on tracks one and three and the second on two and four. The position of those two heads, mounted on a specially designed head assembly unit, is shown in figure 2.10.

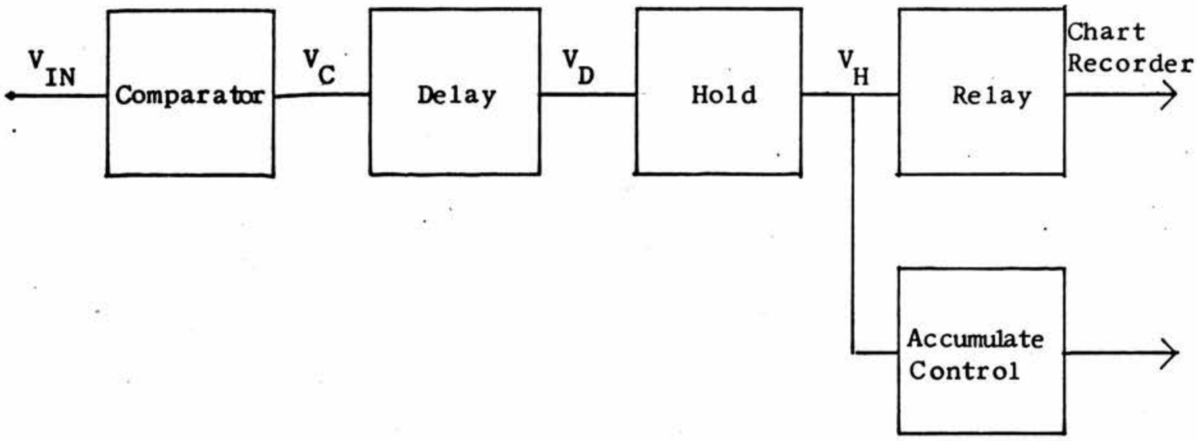
Also shown on the assembly unit is a third head, the advance head. It was so called because if a signal was recorded on channel two or four via the advance head, an advance signal was obtained on replay when using the normal head spacing. Being ahead of the signal on channel one, it was used for triggering an automatic retrieval of the percussion waveform on channel one onto a chart recorder, or in subsequent work into the one hundred channel analyser or the PDP-12 computer.

2.5 AUTOMATIC RETRIEVAL

Automatic retrieval arose out of the need for effortless chart recording. Since percussion sounds were very short compared to the

Figure 2.11 TRIGGER CIRCUIT

block diagram



voltage waveforms

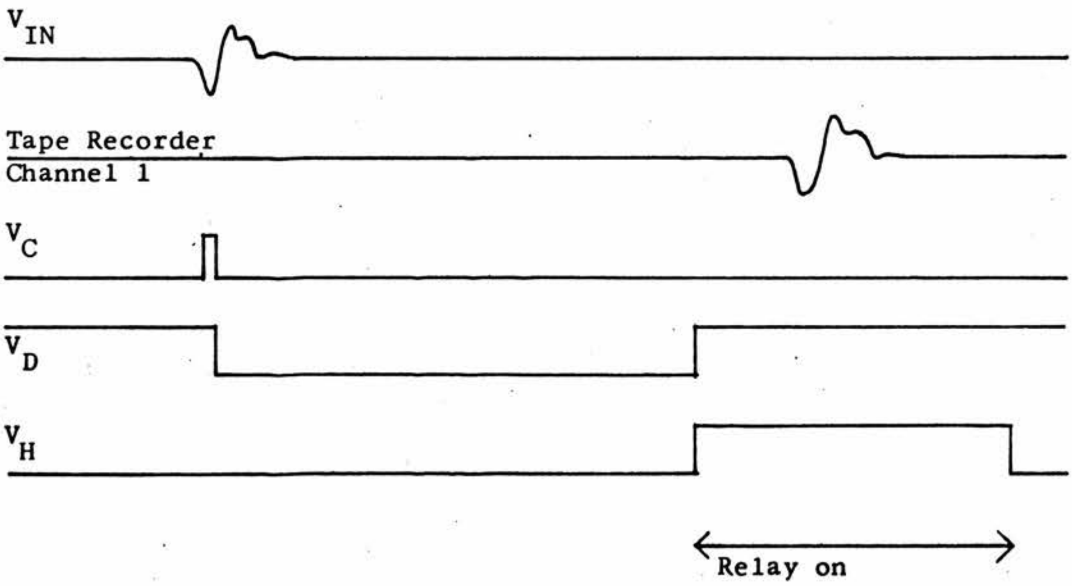
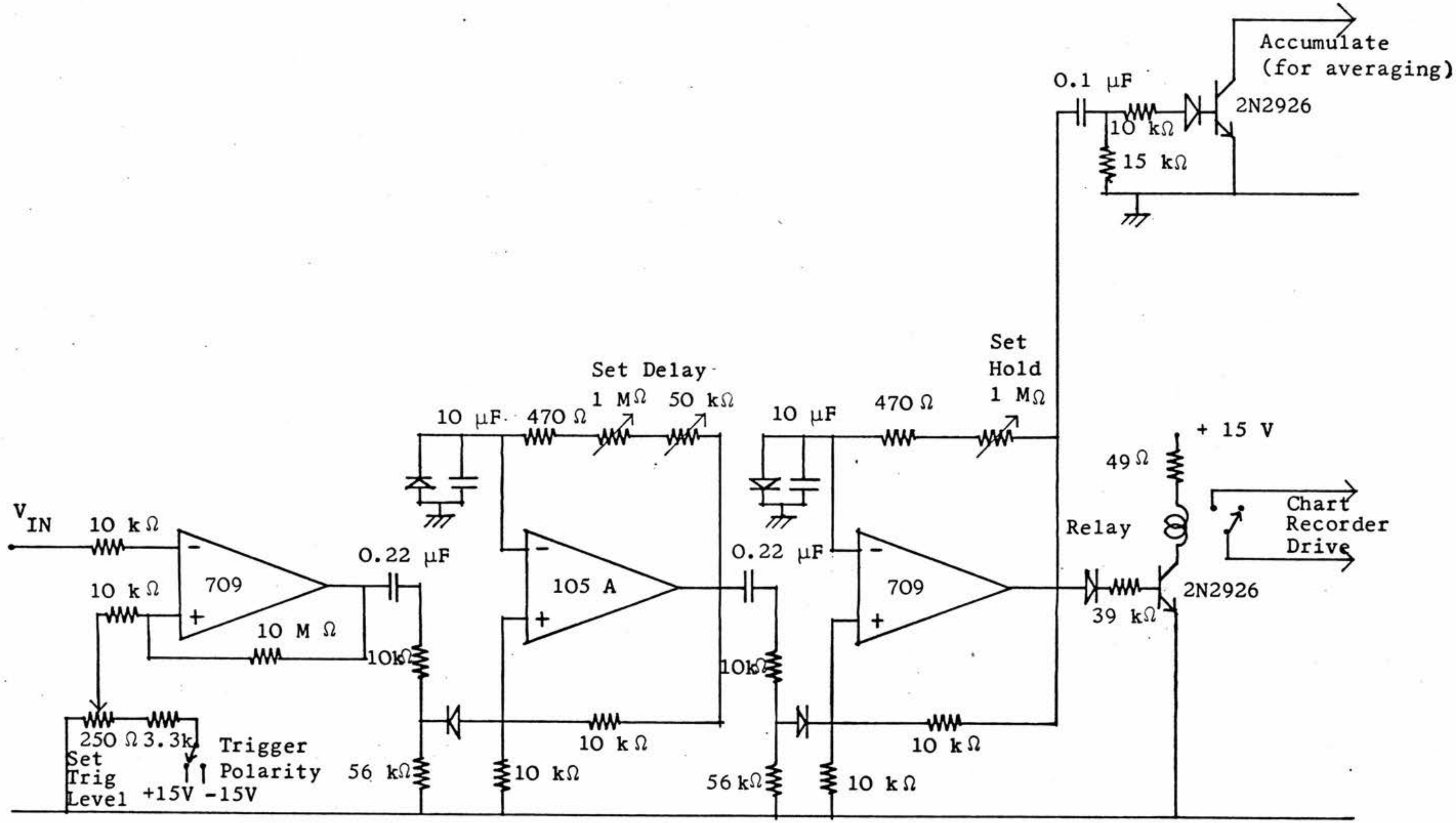


Figure 2.12 TRIGGER CIRCUIT - SCHEMATIC



time between successive sounds, the chart recorder could not be left running continuously, and so the position of each sound on the tape had to be found separately and then charted. Also, if one sound was recorded per second, say, one minute would separate each on replay, making it harder to find each sound. On averaging (see section 4.2) where up to one hundred sounds were averaged, automatic retrieval became an indispensable feature, since averaging required accurate timing of retrieval.

With the advance head lying one inch to the right of the second head, an advance signal four seconds ahead of that on channel one was obtained. This advance signal was fed to the trigger circuit which was programmed to allow the acceptance of all or part of the oncoming signal, by presetting the required delay and hold times.

In figure 2.11 a block diagram of the trigger circuit is shown, and below that, the relevant voltage waveforms to aid an explanation of its operation. The complete schematic diagram is shown in figure 2.12. While the output of channel one was connected to the input of the chart recorder, the output of the channel from the advance head was fed to the trigger circuit.

First of all, the trigger circuit would detect the negative voltage corresponding to the first pressure rarefaction of a percussion sound waveform. This was achieved by the comparator which was preset to detect a certain negative voltage level and when a voltage more negative than that of the preset voltage was applied to the comparator, it would switch and so trigger a monostable controlling the delay. The delay was used to prevent the chart recorder being switched on immediately the advance waveform is detected, because at the slow replay speed a further four seconds would elapse before the percussion sound waveform from channel one could be recorded. Following the 'delay' monostable, and triggered from its positive output voltage step, was another monostable; this one was used to hold on the relay controlling the power to the motor of the chart recorder. The variable hold was preset for extraction of the required length of waveform.

Like the relay and its associated driving transistor, an 'accumulate' control was also connected to the 'hold' monostable. This control circuit was used to initiate an accumulation cycle of the one hundred channel analyser, employed in averaging, by providing a short circuit across the 'accumulate' socket of the analyser. By turning the transistor of the 'accumulate' control on at the beginning of the hold period, a sufficiently low resistance to initiate an 'accumulate' cycle was achieved.

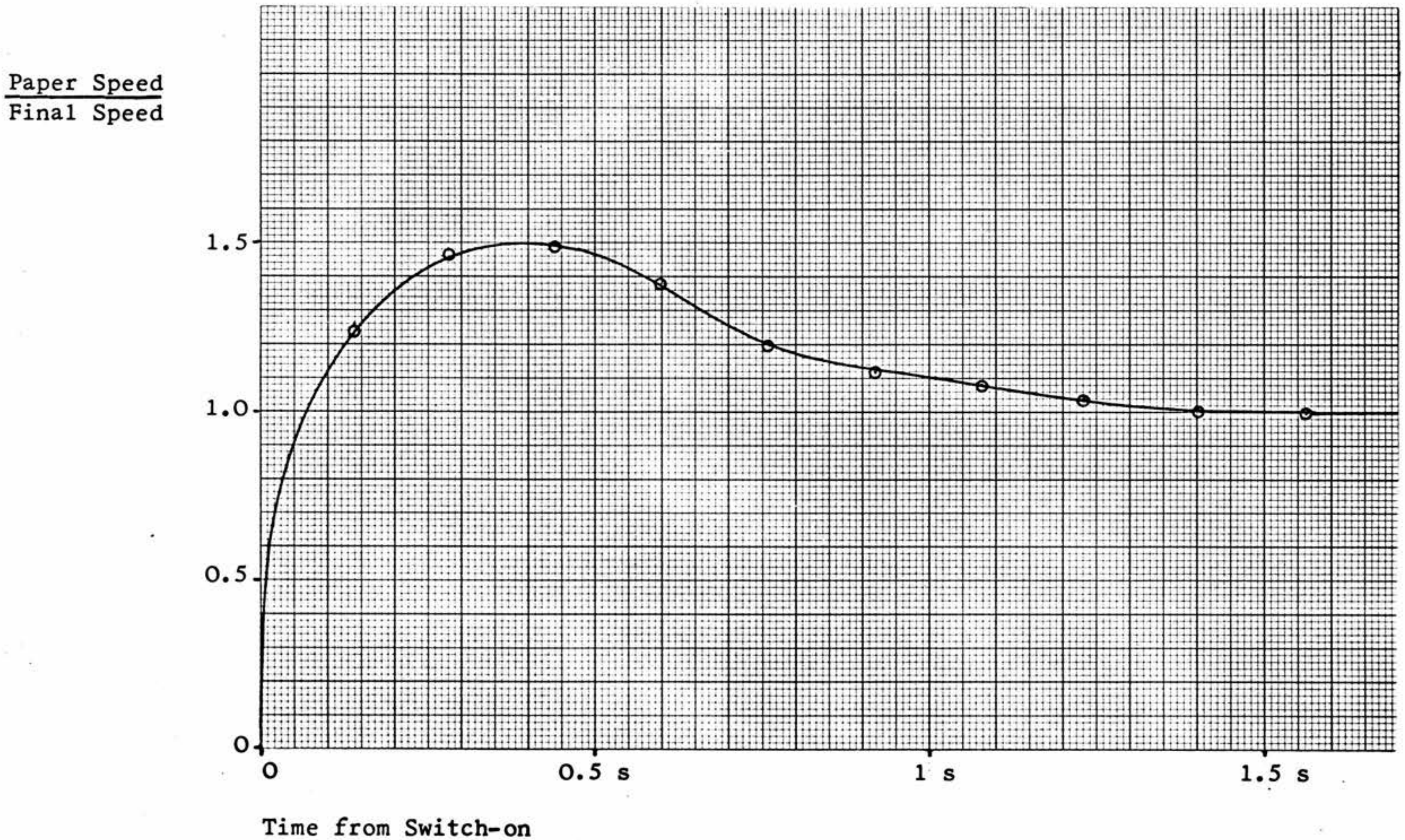
2.6 CHART RECORDERS

Obtaining a permanent visible record is of immense importance when a voltage waveform has to be studied in detail. Modern chart recorders are ideal for this purpose. They contain special paper which is moved at a constant rate past a pen whose deflection on the y-axis is directly proportional to the input voltage; the paper speed being set by a control on the instrument.

The particular recorders employed in this project made use of heat sensitive paper in conjunction with pens having heat elements, thus reducing the pen friction from that of an ink pen and so giving an increased frequency response; in this case 0 to 80 Hz. This was more than adequate for recording percussion sounds on slow replay, where, as already noted, the 1.5 kHz maximum frequency was reduced to 25 Hz.

Figure 2.13 CHART RECORDER PAPER SPEED AFTER SWITCH-ON

One Channel Recorder



Two recorders were used - a single channel (Devices) and a double channel (Devices type M2R).

One feature of particular importance to this project was the time taken to reach, or settle to, the final paper speed after switch-on. This was measured, and it was found that the one channel recorder required 1.5 s as against less than 100 ms, for the two channel recorder, to settle within 1% of the final speed. Figure 2.13 shows how the chart paper for the one channel recorder speeds up after switch-on. At first the quickly increasing speed overshoots to a value 50% greater than its final value before gradually settling to that final speed.

To prevent distortion of the percussion waveform by the chart recorder, no attempt was made to record any waveform until the speed of the chart paper had settled to within 1% of its final value. Therefore when the one channel recorder was in use it had to be switched on at least 1.5 s before the percussion waveform was to be produced. Since the advance head provided for a signal 4 s ahead of that on a normal channel, the delay on the trigger circuit was set at approximately 2 s. When the two channel recorder was used the delay was extended to about 3.5 s because of its superior paper drive system.

CHAPTER 3

REPRODUCTION OF TRANSIENT SOUND PRESSURES

3.1 HEARING AND THE EAR

Medical practitioners at present judge percussion sounds subjectively. Nevertheless they are able to recognise the features of any particular type of sound, and although unable to give an absolute judgment, can assess the relative 'quality' of two sounds. This is analogous to the identification of absolute and relative pitch of a musical note. Hence the ability of the ear and brain to identify percussion sounds presented a suitable means of discovering which features of these sounds the listener considered to be of greatest importance for their identification.

With this end in view it was desirable to be able to listen to various artificial percussion sounds; sounds which had particular features either removed or emphasised so that the relative importance of these features could be discovered. The only practical solution to this problem was to produce the pressure waveform from the equivalent electrical waveform, rather than directly. An added advantage of such a system lay in its ability to reproduce at any required time sounds, which had previously been recorded on magnetic tape.

Accuracy in producing a pressure wave directly proportional to the input voltage was considered to be of great importance. This is another example of where the complete system had to be taken into consideration, so that the sound reproducer would not alter the desired characteristics of the artificial sound waveform, or in the case of real percussion sounds, so that the original sound would be preserved. To preserve the shape of any waveform the phase characteristics of all frequency components must be kept unchanged and so it was desirable for the sound reproducer to have a transfer function which was as flat as possible throughout the frequency range required by percussion sounds.

Much has been said about the ear being insensitive to phase changes, however this is often said in the context of continuous tones where a phase change could equally well be described as a time displacement. The effect of true phase changes on a transient signal can become quite noticeable. For example, an impulse and random noise both have identical amplitude-frequency characteristics but vastly different phase-frequency characteristics, and needless to say, both sound quite different to the ear. That example, although of an extreme nature, does show that phase and waveform changes can be important, and so further emphasises the need for the sound reproducer to conserve the input waveshape as far as possible.

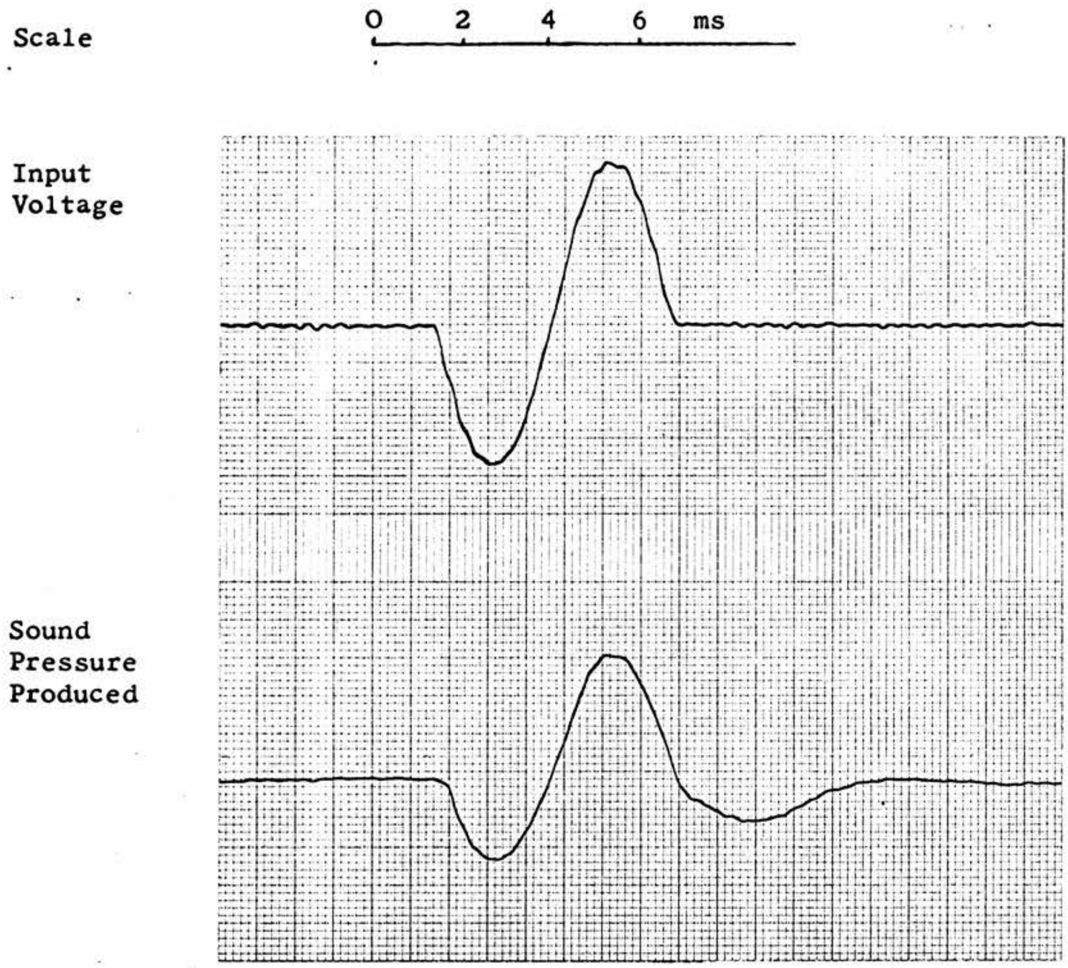
3.2 SOUND PRESSURE REPRODUCER

3.2.1 Previous attempts

Attempts at producing a pressure wave from an electrical signal presented a number of difficulties, because the sound pressure was required to accurately follow the input voltage. The main difficulty in reproducing sounds arises due to the cone mass of the reproduction system, which causes the cone position to lag behind the input voltage; this lag becoming especially noticeable on short transients. After the input voltage has returned to zero, the cone continues to move under the energy of the remaining inertia, resulting in an overshoot or, if the system is underdamped in damped oscillation.

Loudspeakers with a heavy cone always exhibit damped oscillation

Figure 3.1 TRANSIENT RESPONSE OF HIGH-QUALITY MOVING COIL HEADPHONE



after a transient input voltage, because it is impossible to damp the cone sufficiently, without seriously degrading the loudspeaker's high frequency response. An unsuccessful attempt was made by Casteleijn (1961) when he tried to reproduce percussion sounds using a loudspeaker.

The result of another attempt, this time using high-quality moving coil headphones is shown in figure 3.1. Less than one gram of paper had been used for the cone and yet the mass was too great for reproducing transient sounds; the effect of the pressure lagging the voltage input can clearly be seen on the initial response, and the system although damped sufficiently to prevent oscillation still exhibited the overshoot due to the remaining energy of the cone.

3.2.2 Proposed technique

If any sound reproducer is to be used at all, it must be of the headphone type where the pressure changes, formed in an enclosed space, are directly proportional to the diaphragm displacement. The loudspeaker, on the other hand, by radiating the sound, does not produce a pressure wave which is related in any simple way to its cone displacement or electrical input signal. This point can be illustrated by referring to the case of the vibrating sphere (Appendix A.1). The sphere, unlike the loudspeaker cone, lends itself to a relatively simple mathematical description and yet, even there, a differential equation must be solved before the resulting pressure can be determined.

As previously noted, the main requirement of the headphones is that of low cone or diaphragm mass, although diaphragm compliance and damping are also important in preventing the diaphragm vibrating at its natural frequency. Today, light strong diaphragm materials, such as polyurethane sheet, are now available with a mass comparable to that of a film of air a few millimetres thick. Some such light diaphragm used in conjunction with an electrostatic technique has enabled sound reproducers with unusually smooth pressure responses to be developed. Other advantages of the electrostatic technique arise from the uniform pressure exerted over the diaphragm by the electrostatic force, which lessens any tendency for natural modes of vibration to be set up on the diaphragm itself, and from the removal of the need for a heavy voice coil. Although electrostatic transduction is by no means new, only recent advances in engineering techniques and materials have enabled practical units to be produced.

3.2.3 Electrostatic Transduction

All electrostatic transducers, without exception, employ a high electrical potential on the diaphragm. This polarisation potential induces a charge on the diaphragm due to the capacitance formed between the diaphragm and the two fixed perforated plates surrounding it. A full theoretical treatment has been made by Hunt (1964), but all that need be understood here is that the greater the charge on the diaphragm the greater will be both the force acting on it and the sound pressure resulting from a given signal voltage. Hence the need for a polarisation potential.

3.2.4 Electrostatic Headphone System

A pair of electrostatic headphones (Stax SR-3) were purchased. However, the input voltage to these headphones was coupled to each earpiece through a transformer, which determined the low frequency cut-off; 30 Hz in this case. This was too high for preserving the waveshape, so a directly coupled amplifier was developed to power them.

Figure 3.3 ELECTROSTATIC HEADPHONE AMPLIFIER

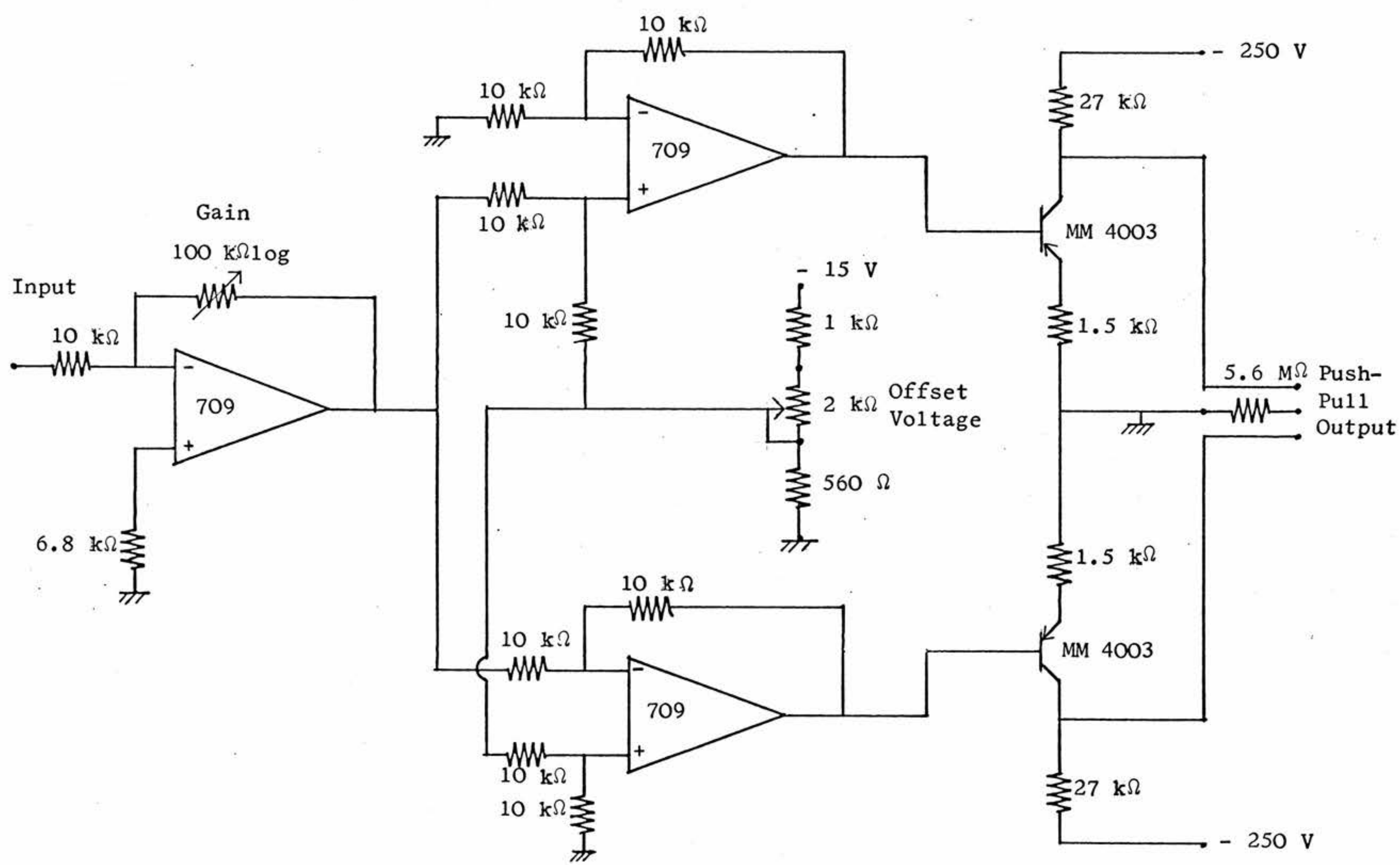


Figure 3.2 BLOCK DIAGRAM OF HEADPHONE AMPLIFIER

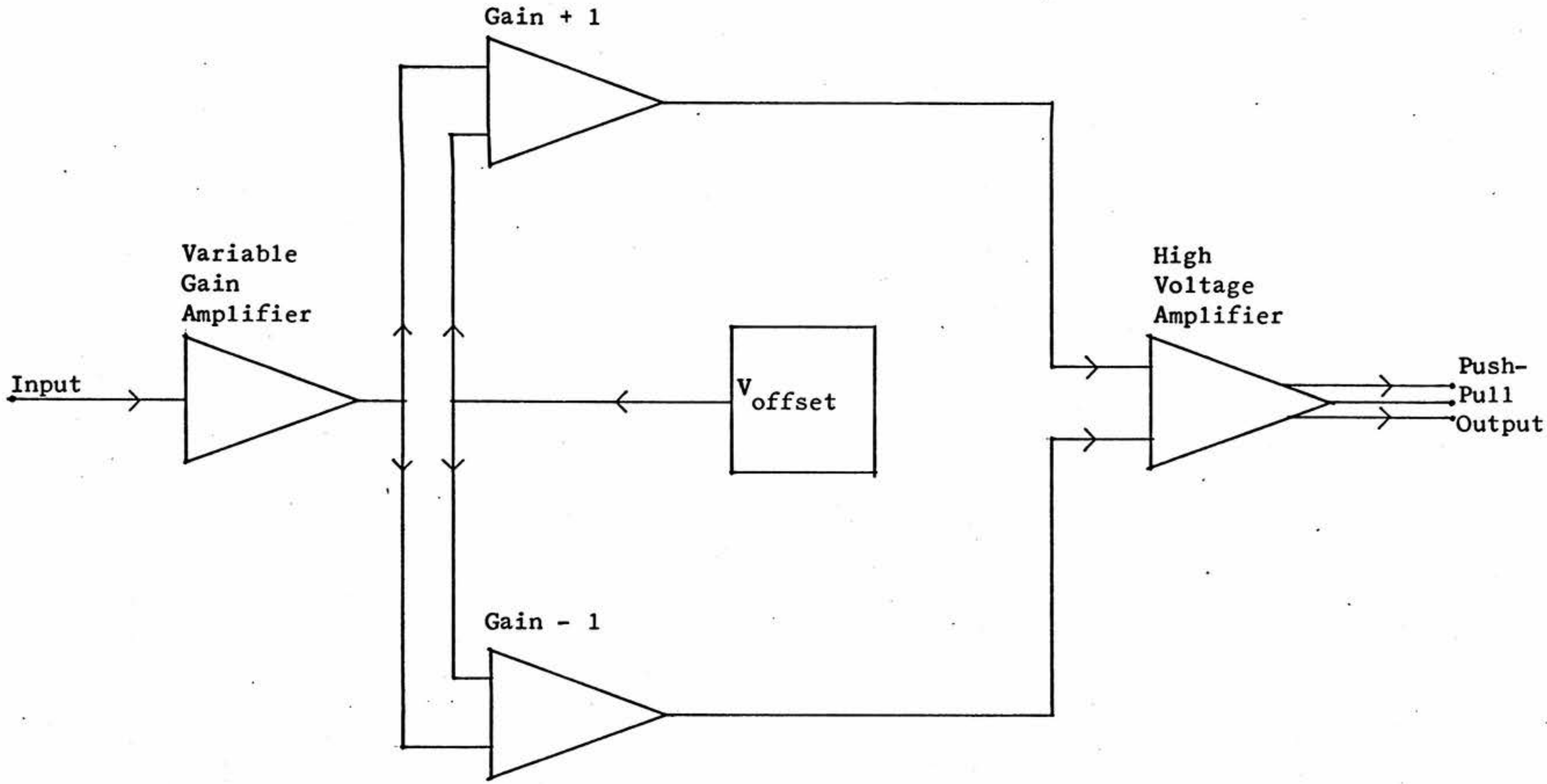


Figure 3.4 MICROPHONE-HEADPHONE COUPLER

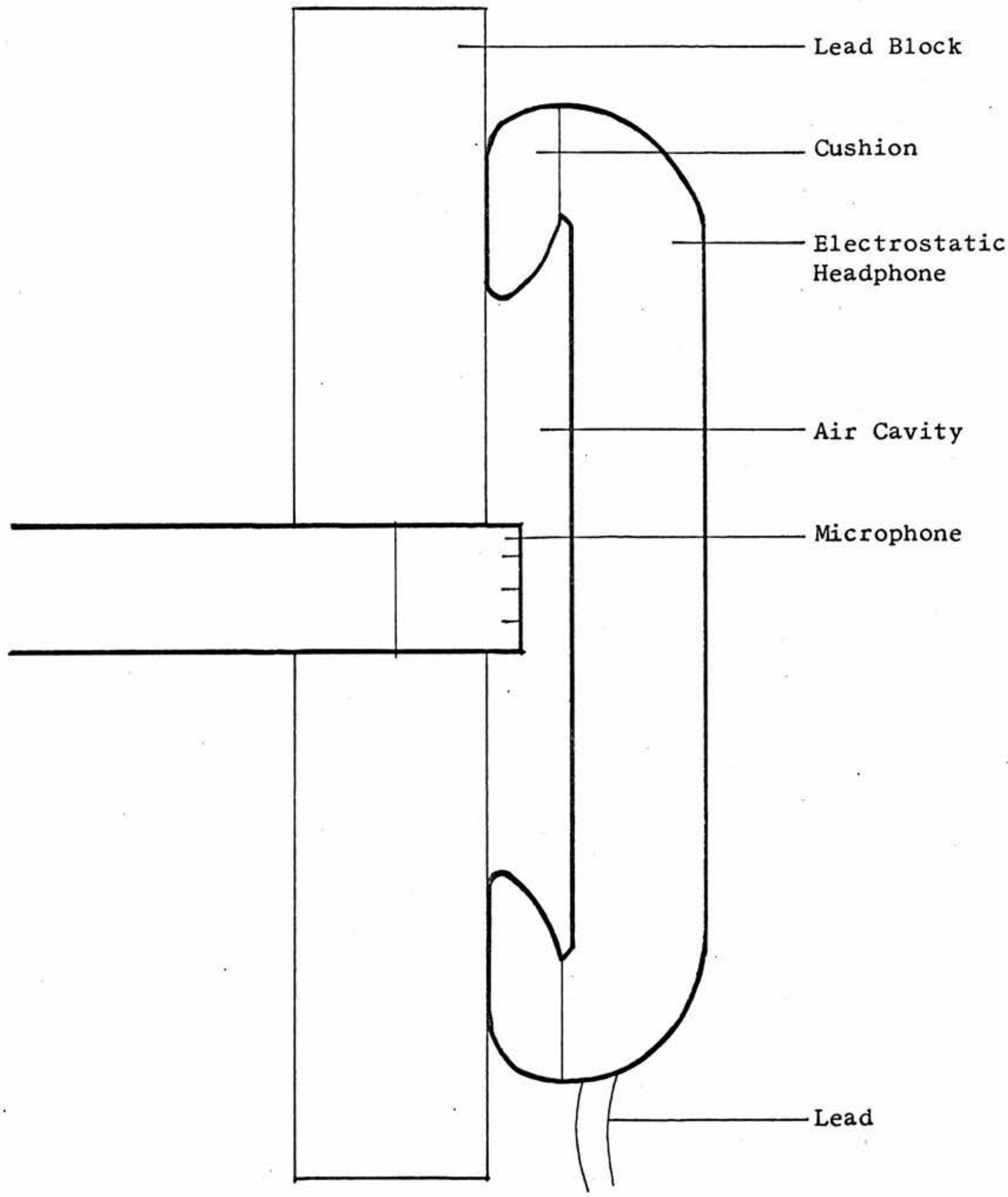
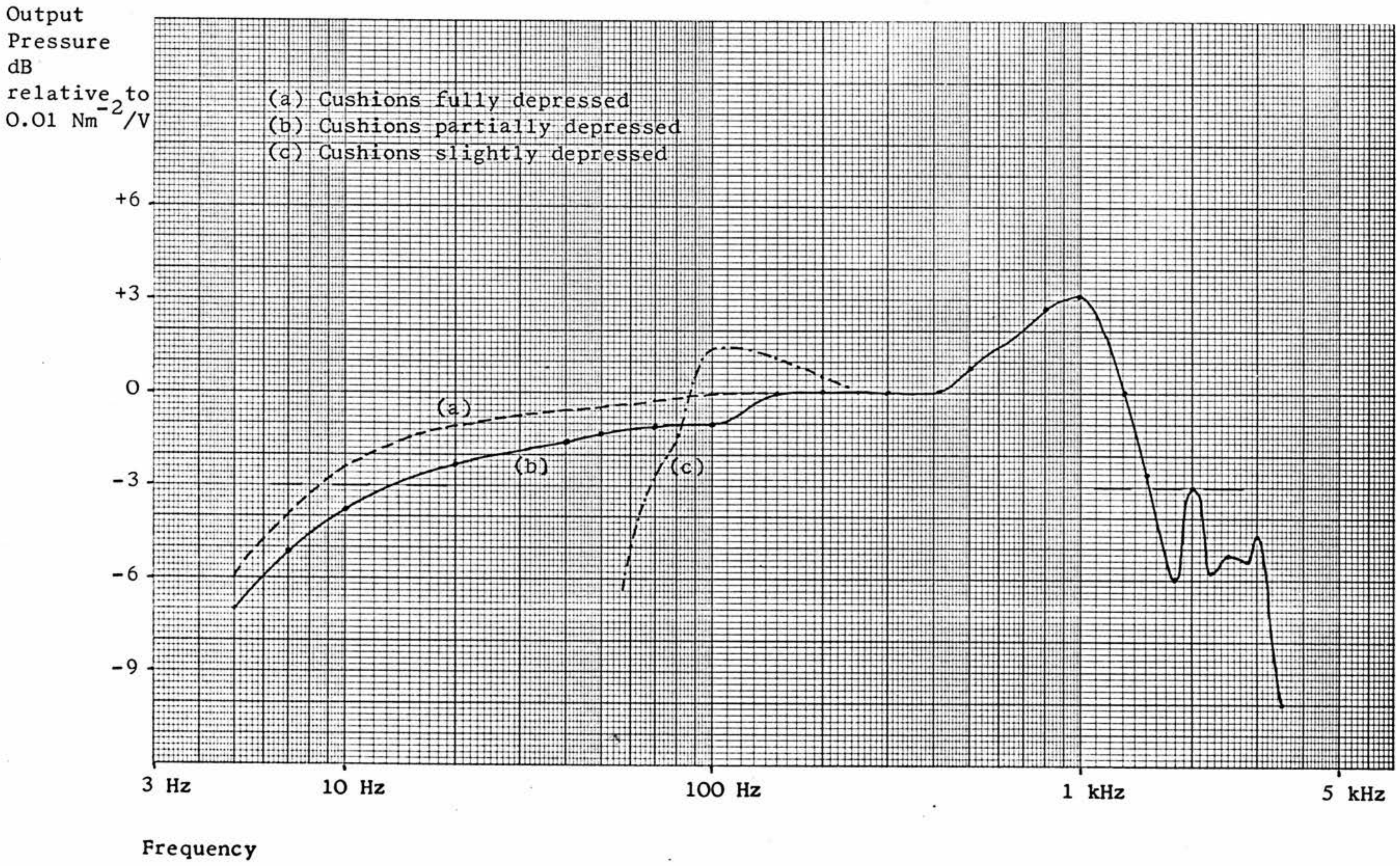


Figure 3.5 PRESSURE RESPONSE OF ELECTROSTATIC HEADPHONES



The amplifier was designed to give a push-pull output centred at 180 V and so at the same time quashed the need for a separate high polarisation voltage supply. Shown in figure 3.2 is the amplifier block diagram and in figure 3.3, the schematic diagram.

With the headphones coupled to the B. and K. 4117 crystal microphone to measure the headphone frequency response (figure 3.4) it was found that the coupler added its own response to that of the headphones. A number of different couplers were tried before the final version was made from lead. Wood had been used for the first coupler, but natural resonances of the wooden block fell within the frequency range being measured. Lead was found to be the best material tried as the first spurious response of the lead block fell higher in frequency than any of the other couplers and as the cut-off frequency fell at 1.5 kHz (figure 3.5) it was not felt to be worthwhile expending any additional effort on the coupler since 1.5 kHz had previously been determined as being the upper limit required for reproducing percussion sounds.

The cushions surrounding the earpieces were also found to add their characteristics to the pressure response, this time at lower frequencies. At these low frequencies the pressure tended to be kept constant by the cushions being compressed to take up any tendency for a decrease in pressure and vice versa. Curve (c) of figure 3.5 is the result obtained with the cushions very lightly depressed, and shows that most of the low frequency energy had been absorbed. With the cushions fully depressed their influence was completely removed from the response (curve (a)) giving a low frequency cut-off of 9 Hz; this cut-off being determined by the inbuilt leakage of the headphones. Curve (b) was

Figure 3.6 TRANSIENT RESPONSE OF ELECTROSTATIC HEADPHONES

Scale 0 2 4 6 ms

Input Voltage

Sound Pressure Output

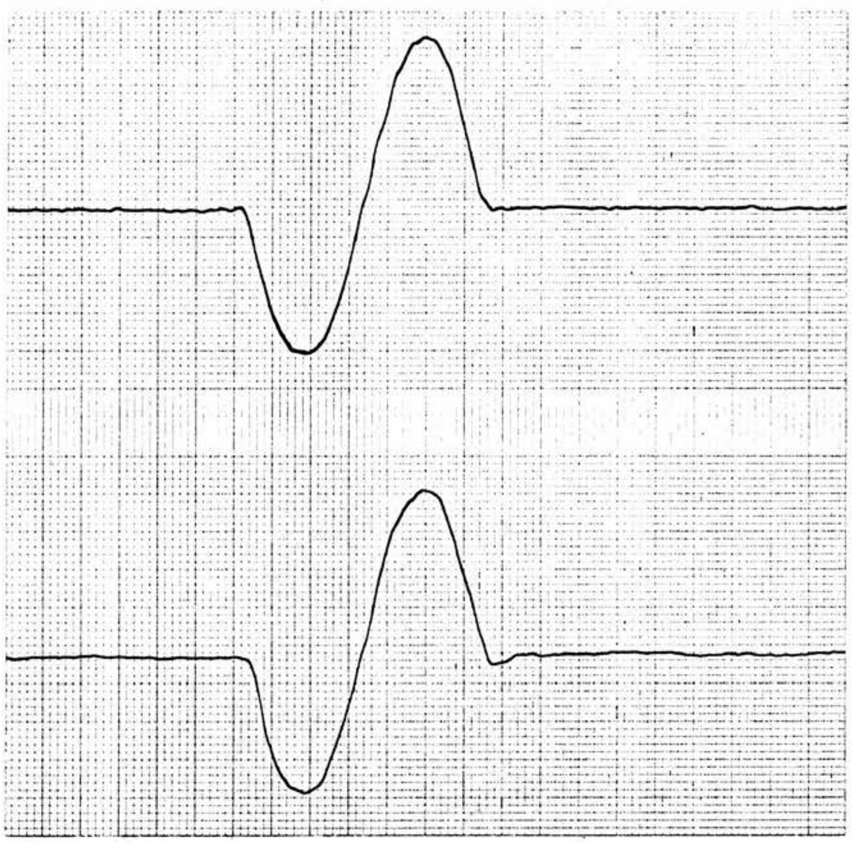
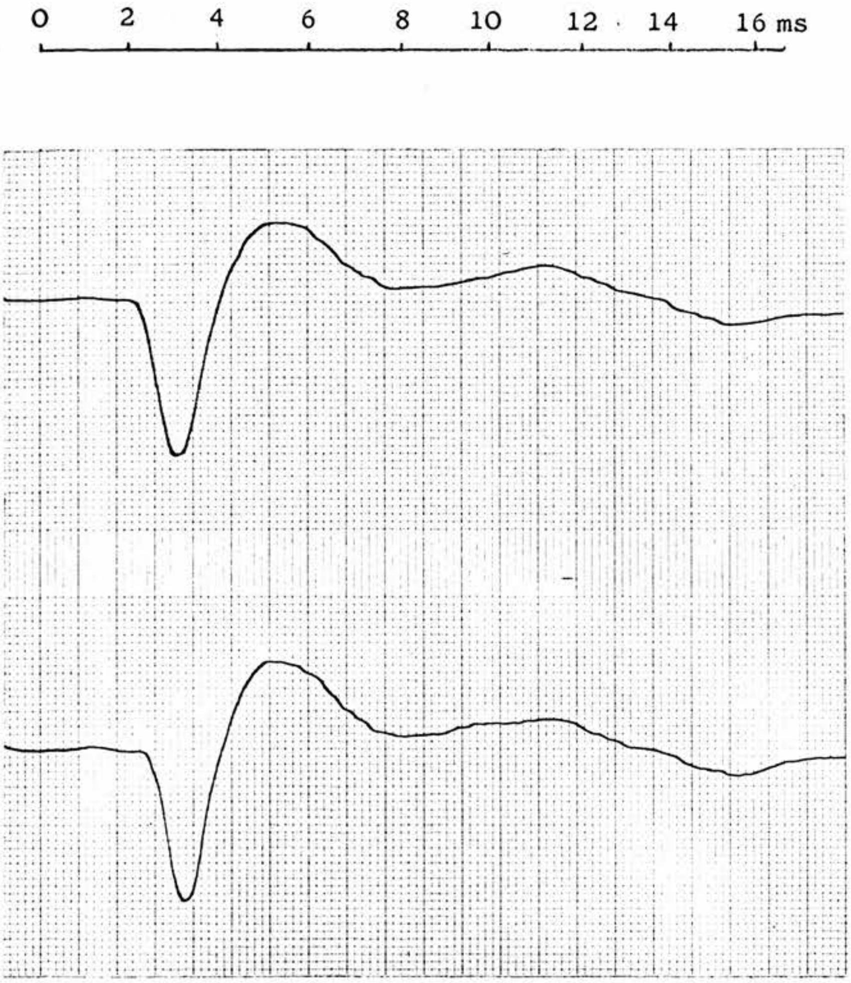


Figure 3.7 REPRODUCTION OF A PERCUSSION SOUND WITH ELECTROSTATIC HEADPHONES

Scale



Sound Pressure Recorded by Microphone

Sound Pressure Output

obtained with the cushions partially depressed.

With an understanding of the features determining the frequency response of the headphones, they were then tested for their response to a transient waveform similar to that used in testing the high quality moving coil headphones producing the response of figure 3.1. Negligible overshoot occurred this time (figure 3.6) confirming their excellent transient response. No percussion waveshape demands quite so abrupt pressure changes, but nevertheless the reproduction of an actual percussion sound was tried. This was especially important since the low frequency response of the reproduction system did not extend down to the 3 Hz necessary for retaining the original waveshape, and so this served as a first check of the system with actual percussion sounds. Figure 3.7 illustrates the result obtained on reproducing a typical 'resonant' sound; the top trace being a recording of the original sound pressure and the bottom trace, that of the sound reproduction. Although not conserving the waveshape perfectly, the resulting sound pressure waveform closely followed the original waveshape. (It should be noted that both figures 3.6 and 3.7 were recorded with the cushions semi-depressed.)

Finally, the headphones were subjected to listening tests and it was found that the reproductions were unmistakably true to their original sounds since the noticeable types of distortion such as damped oscillation or overshoot were not generated in this system. As far as the ears were concerned there was no audible difference between the sounds reproduced with the cushions either semi-depressed or with the partial depression afforded by the headband connecting the earpieces.

Therefore the headphones were able to be used when required with a knowledge that they were reproducing the sounds intended.

CHAPTER 4

QUALITATIVE ANALYSIS OF PERCUSSION SOUNDS

4.1 INTRODUCTION

By far the greatest effort expended by research workers into the investigation of percussion sounds has been directed towards obtaining a purely subjective description or analysis of the sounds. This has led to numerous ill-defined descriptions. One particular example of this arose in Skoda's (1839) work. When describing the sounds by means of his four 'qualities', he used the terms 'full' and 'empty', and since then, as noted by Delp (1968), both have been a matter of much dispute.

Subjective analysis was abandoned in this present investigation. This chapter deals with the qualitative descriptions, preparing the way for, in later chapters, a quantitative description.

Nevertheless, since subjective descriptions are the only ones applied by the medical profession at present, they were not totally disregarded. Some of the results of the work carried out were related where necessary to the existing descriptions. For example, results from Fourier spectrum analysis were correlated with those descriptions which have attempted to assess by ear the frequency content of the various sounds.

4.2 OBSERVATIONS ON PERCUSSION SOUND PRESSURE WAVESHAPE

If percussion sounds were not to be analysed subjectively, how were they to be analysed?

These sounds reach the medical practitioner's ear through the air, and it is the pressure changes in the air which carry the audible

Figure 4.1 ANATOMY OF THORAX AND ABDOMEN

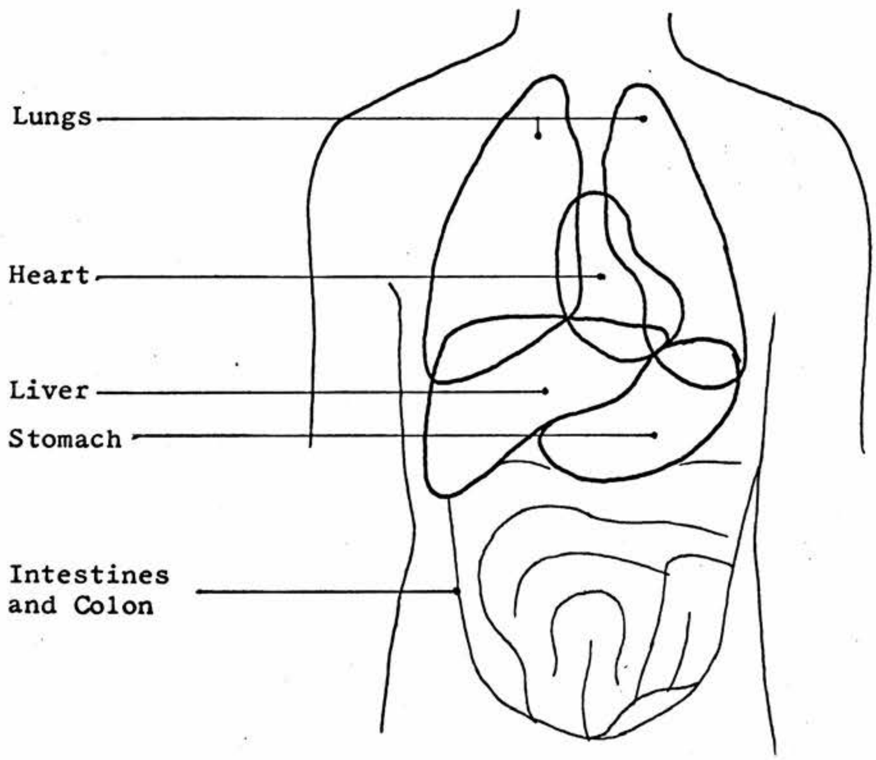
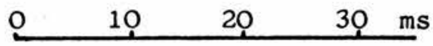
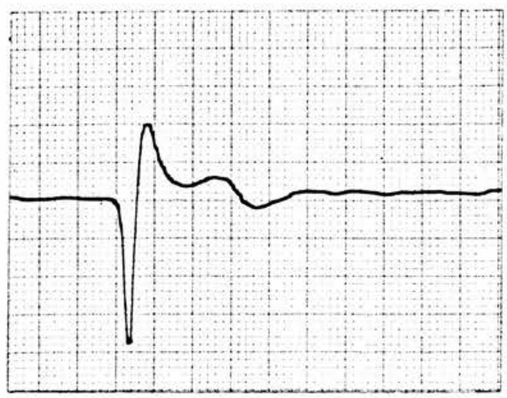


Figure 4.2 TYPICAL PERCUSSION SOUNDS

Scale



'Resonant'



'Dull'

'Tympanic'

information. Hence, as a starting point for this investigation, it was decided to study the sound pressure waveforms corresponding to each type of percussion sound.

With the aid of the recording system, numerous pressure waveforms were obtained to illustrate the various percussion sounds. Although no attempt was made at this stage to formulate a description of the sounds, their salient features were noted.

Sounds from the healthy human body were dealt with, since these sounds had to be understood first before any deviation from the normal could be detected. In any case, sounds similar to those generated over diseased organs can often occur in the healthy body. For instance, a lesion in the lung can have the same effect as the liver when it underlies the lower region of the lung.

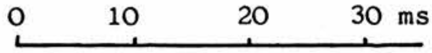
Before discussing the sounds it is interesting to note briefly the regions from which particular sounds emanate. Those organs of greatest relevance to percussion are outlined on a simplified diagram of the human thorax and abdomen in figure 4.1. Percussion is mostly used in the examination of the lungs, over which a 'resonant' sound would be expected. The liver or heart, when they underlie the lung, decrease the 'quality of resonance'. When no lung is present, both these organs produce a 'dull' sound. 'Tympanic' sounds are produced over the stomach, colon and intestines.

The three sounds just referred to - 'resonant', 'dull' and 'tympanic' - are the most distinctive ones. Waveforms of each type are shown in figure 4.2 'Resonance' was recorded over the right lung, 'dullness' over the liver, and the 'tympanic' sound over the abdomen.*

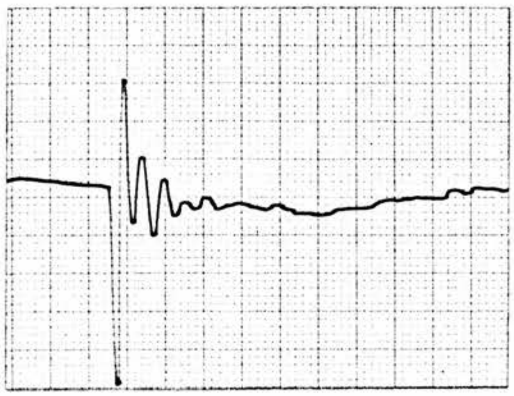
* To aid standardisation, these sounds as well as all others in this thesis have been recorded, unless otherwise stated, with the microphone positioned at approximately 15 cm from the chest. In addition, the microphone was placed to lie along the perpendicular from the percussion point. The chart recordings obtained show a full scale sound pressure variation of approximately $\pm$ 0.3 Nm^{-2}

Figure 4.3 'TYMPANIC' SOUNDS

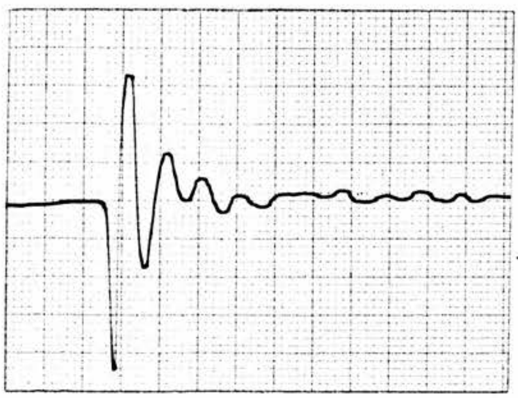
Scale



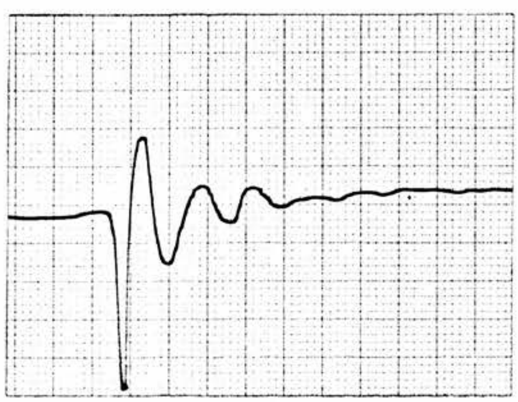
Stomach



Abdomen -
Right Lateral
Region



Abdomen -
Right Inguinal
Region



Although all three contain a sharp initial rarefaction pressure spike, a marked contrast can be noted between subsequent parts of the waveshapes. The large compression peaks of 'resonance' clearly distinguish it from the poorly defined variations of 'dullness'. Contrasting with both is the 'tympanic' sound. Its waveform, not dissimilar to a damped sinusoid, shows why it has been described as having a more definite pitch than any other percussion sound.

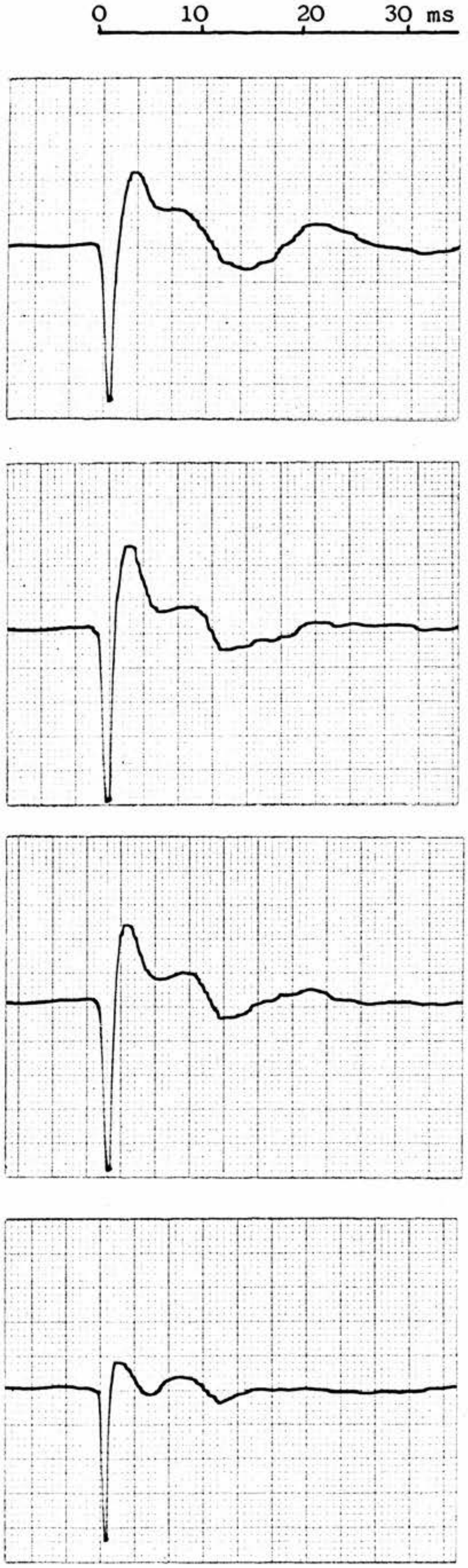
Even on one subject, the 'frequency' of the damped sinusoid of the 'tympanic' sounds varies considerably. Illustrative of this variation are those sounds shown in figure 4.3. All were taken from the same subject, but over different areas of the body - one over the stomach and the other two over different positions on the abdomen.

The term 'frequency', however, should be restricted to the description of continuous periodic variations. Further consideration will be given to this in the section on frequency analysis. Nevertheless, a measure of frequency was obtained by averaging the period between successive damped cycles of the 'tympanic' sounds. Over the stomach a 'frequency' of approximately 600 Hz was measured; and over the abdomen, 300 Hz at the right lateral region and 200 Hz at the right inguinal region. Those figures must in no way be considered as 'normal' for various regions of the body. They are included to illustrate typical variations in 'tympanic' percussion sounds.

Other sounds do not lend themselves to such a neat description as those of 'tympanic' quality. 'Resonance' and 'dullness', although they both have their own distinctive and easily recognisable type of waveform, evade a simple description. Hence the bulk of this present thesis deals with non-'tympanic' sounds.

Figure 4.4a VARIATION OF SOUNDS ON CHEST

Scale



2nd i.c.s.

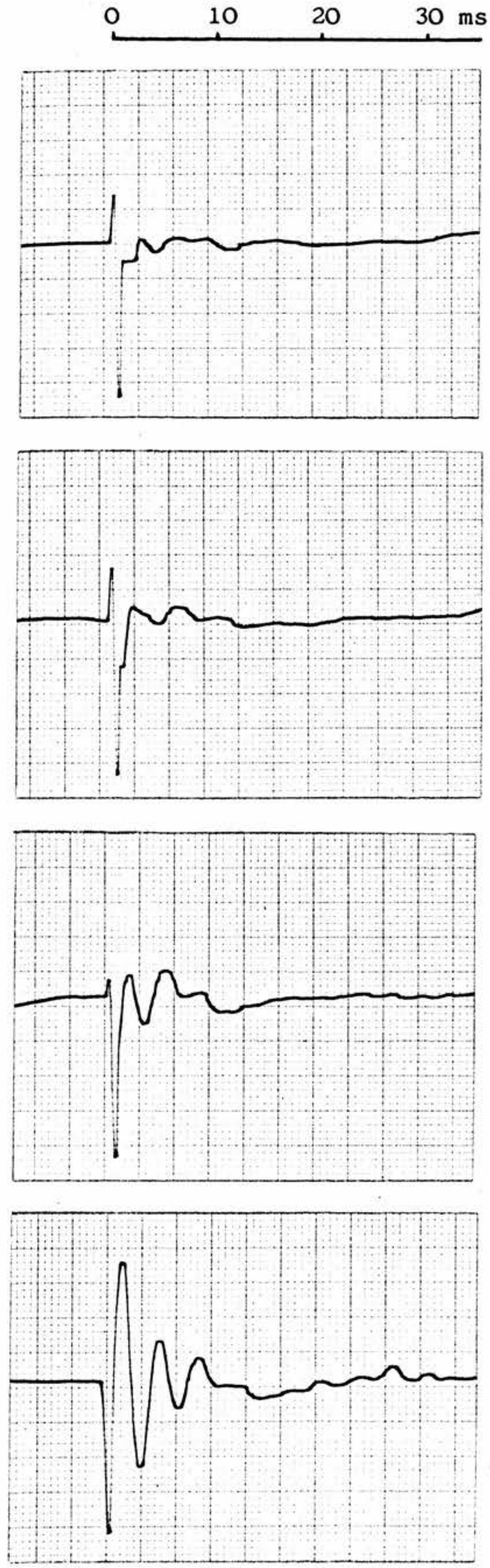
3rd i.c.s.

4th i.c.s.

5th i.c.s.

Figure 4.4b VARIATION OF SOUNDS ON CHEST

Scale



6th i.c.s.

7th i.c.s.

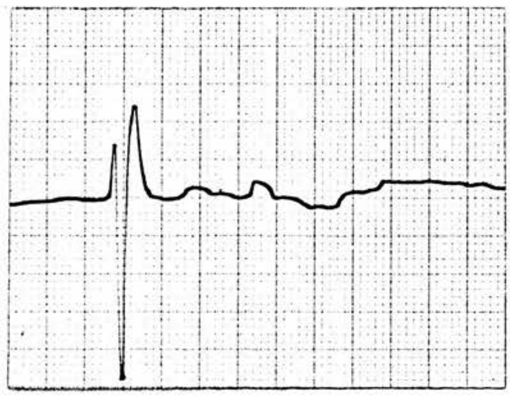
8th i.c.s.

Sub costal

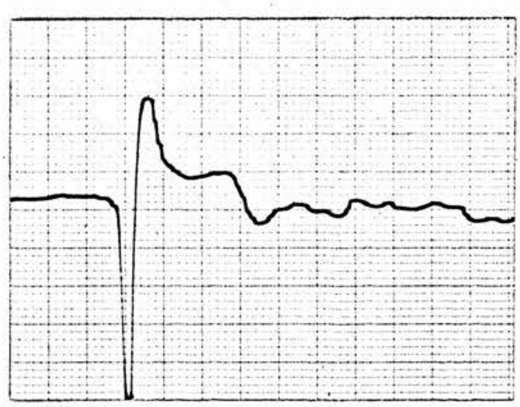
Figure 4.5a EFFECT OF LUNG VOLUME

Scale 0 10 20 30 ms

RV



RV + 1 litre

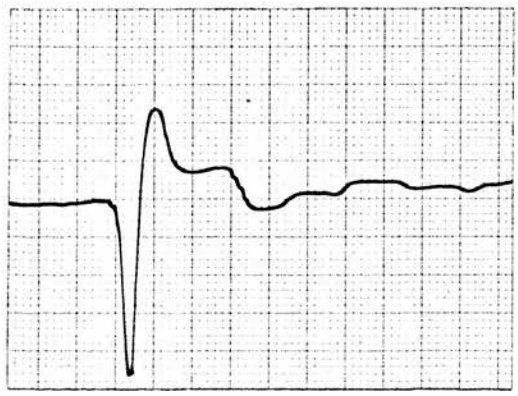


RV + 2 litres

Figure 4.5b EFFECT OF LUNG VOLUME

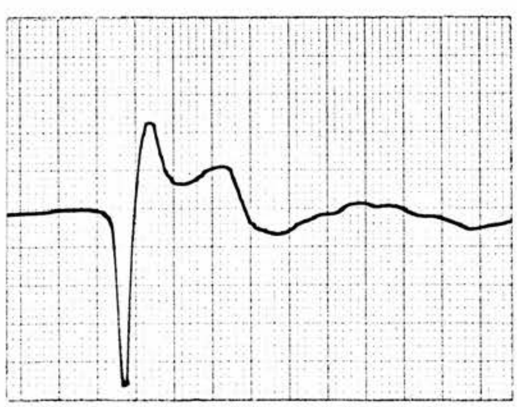
Scale 0 10 20 30 ms

RV + 3 litres



RV + 4 litres

RV + 5 litres



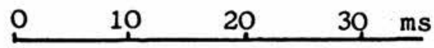
Several factors can complicate matters by introducing large variations in the waveform, even on a normal healthy subject. For example, lung volume and the percussion position over the lung both play a noticeable part in altering the waveshape.

Consider the position of percussion first. Figure 4.4a, b illustrate typical variations in the sound pressure waveform obtained by percussing on the right-hand side of the chest with the lung volume at its functional residual capacity (F.R.C.). Sounds from the second, third and fourth intercostal spaces (i.c.s.) all exhibit characteristically resonant features, even though these features can be seen to vary. Over the fifth intercostal space the sound has lost a great deal of its 'resonance' because only the base of the lung had been lying immediately below. Lower down the chest still, the sound has become 'dull'. With no lung underlying those areas, all trace of a 'resonant' quality has been lost. However the waveforms do contain an other feature which is especially prominent at the eighth intercostal space. This is the 'tympany' of the abdomen which had been excited by the percussion blow. Moving off the rib cage altogether on to the subcostal region, the truly 'tympanic' note was obtained.

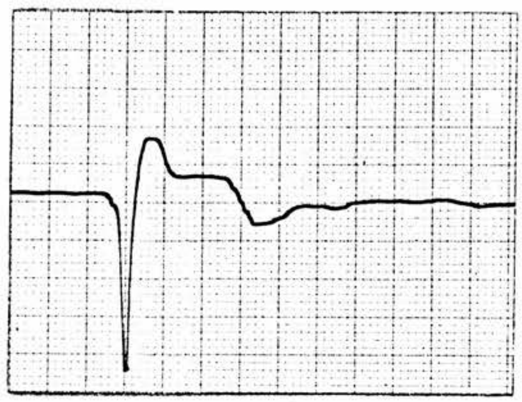
Like position, the volume of air contained in the lungs played a large part in determining the quality of the 'resonant' sound. Figure 4.5a,b show the typical changes in waveshape obtained when the total lung volume was increased from the residual volume (RV), at full expiration, in steps of one litre to a maximum of five litres above the residual volume. This was the subject's total lung capacity (T.L.C.). These volumes were measured with a spirometer, and the percussion sounds were obtained over the right lung at the fourth intercostal

Figure 4.6   EFFECT OF LUNG POSITION

Scale



Full
Inspiration



Full
Expiration

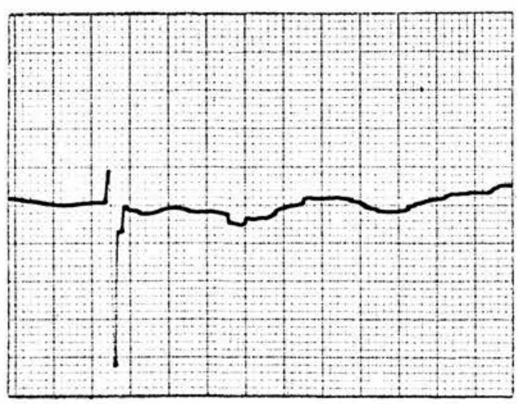
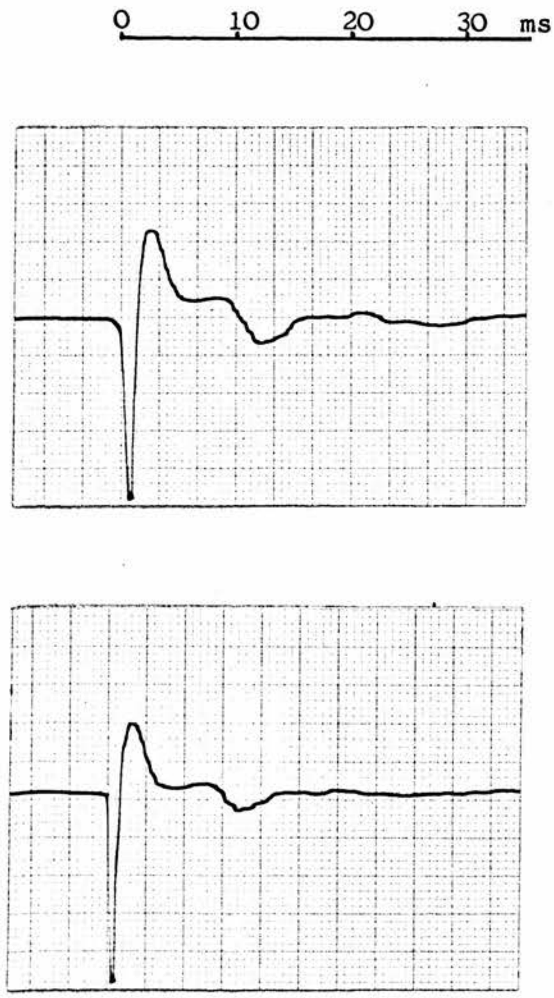


Figure 4.7 EFFECT OF PERCUSSION FORCE

Scale



Normal Percussion

Heavy Percussion

space. Lower volumes, as can be seen, do not display quite so prominently, the characteristic features of the 'resonant' sound at normal inspiration levels. Although they are indisputably 'resonant', they do have a lower 'quality of resonance' It can also be noted that the most marked change in waveshape occurs at the lower lung volumes.

Those differences observed above, due to variation in lung volume or to altering the position of percussion on the chest were later quantitatively measured (see Chapter 7). This enabled the small changes in sound to be more easily studied.

On some positions of the thorax the percussion sound can vary naturally between 'resonance' and 'dullness'. This occurs over those areas where, depending on the phase of respiration, the lung may or may not underlie. Both traces of figure 4.6 were taken over the same spot on the fifth intercostal space; the upper trace showing a 'resonant' sound on full inspiration, and the lower, a 'dull' sound on full expiration.

Another factor - the force of the percussion blow - must also be taken into account. Figure 4.7 compares two sounds generated with different percussion forces.(The pressure scale of the wavetrace with heavy percussion has been contracted to allow for twice the pressure change of the upper waveform.) It should be noted that although the relative size of the features shown were different, the characteristic shape of those features was retained.

4.3 EXTRACTION FROM NOISE BY AVERAGING

4.3.1 Noise

Before any further progress could be made in describing the percussion sounds, the effect of noise had to be considered. This was to ensure that no significant features had been overlooked by being masked in noise.

Noise is here used to refer to everything on the final waveform which was not included in the original sound being transmitted directly from the patient's chest to the physician's ear or, in this case, the microphone of the recording system. Two main categories of noise were present; one was electrical noise generated within the recording system and the other was acoustic noise picked up by the microphone. Included within this second grouping were background room noise and reflections of percussion sounds from hard surfaces such as walls, benches and equipment.

Having established (section 2.4.2) that noise performance of the experimental set-up was determined mainly by background room noise, any improvement had to be made outwith the recording system.

Fortunately background noise, as well as electrical system noise, can be minimised by averaging a number of identical sounds. If all the sounds to be averaged are added coherently, the value of the averaged waveform increases in direct proportion to the number of averages taken. Noise, being random, tends to average out. If truly random, the value of the noise pressure increases as the square root of the number of averages. Hence the signal-to-noise ratio would also increase as the square root of the number of averages.

A theoretical analysis has been carried out by Ernst (1965) for a number of different noise spectra. No matter what the noise spectrum was there was always an improvement in signal-to-noise ratio with averaging.

To confirm that there would be a substantial improvement when

Figure 4.8 AVERAGING OF NOISE SOUND PRESSURE

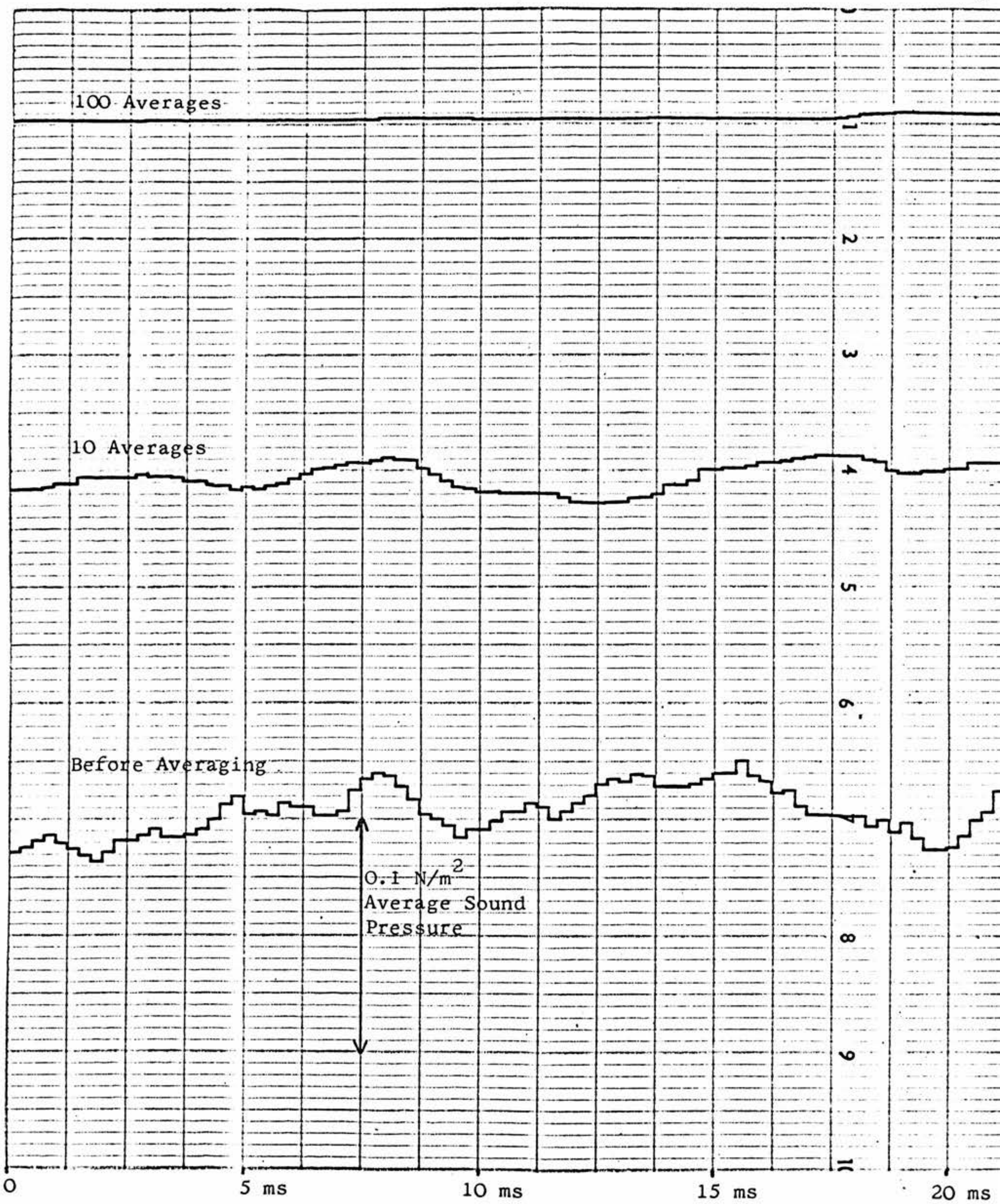
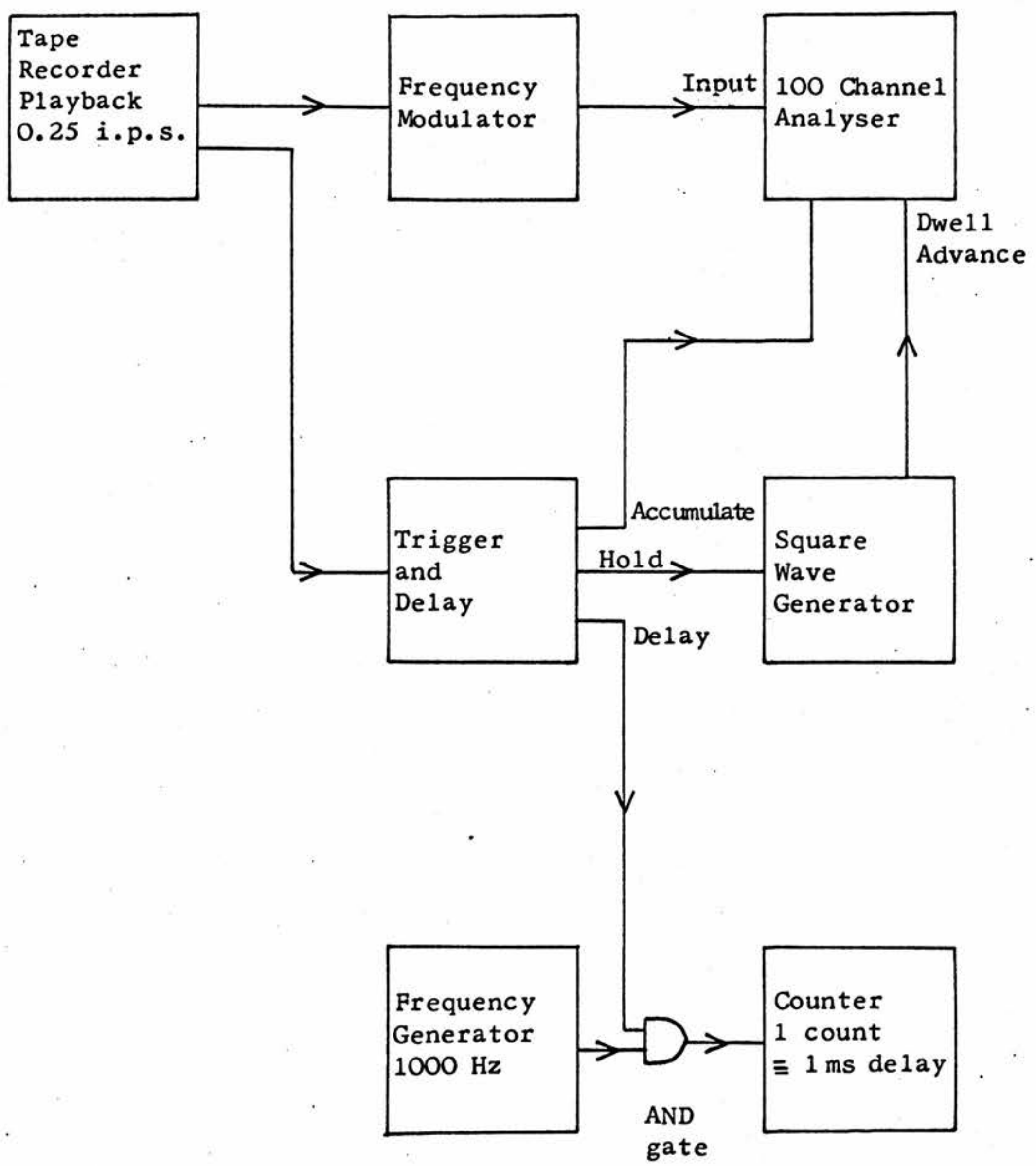


Figure 4.9 AVERAGING SYSTEM



sound pressure noise was averaged, quiet room noise was recorded. Ten and one hundred averages were then taken using the system to be described. The results shown in figure 4.8 have been corrected for average sound pressure. Without doubt, an improvement had been obtained.

To calculate a figure for the increase in signal-to-noise ratio to be obtained, a print out of the sampled values before and after averaging was obtained. From each group of samples the standard deviation was calculated and converted to an equivalent r.m.s. sound pressure. (Sound pressures are measured about the mean pressure.) This procedure was repeated five times in all for each of the three cases and the mean r.m.s. noise pressure calculated. The table below gives the values obtained.

Number of averages of Quiet Room Background Noise	Equivalent r.m.s. Sound Pressure Noise Nm^{-2}
Before averaging	0.0093
10	0.00325
100	0.00125

From the results in the above table, the expected improvement in signal-to-noise ratio was seen to approach closely to the ideal $\sqrt{n}$, where n is the number of averages taken. The improvement was greater than $0.9\sqrt{n}$.

4.3.2 Averaging System

The most important part of the averaging system (figure 4.9) was a one hundred channel analyser (Gammascope Model 102) which was set up in its 'scaler mode' to sequentially sample each incoming waveform one hundred times.

Since the analyser in its 'scaler mode' operated by counting pulses, the analogue waveform was not directly acceptable. A frequency modulator was used to interfacd the tape recorder with the analyser. As each output cycle of the modulator was counted for a preset time by the analyser, this resulted in the number of counts in each channel being directly proportional to the average analogue voltage presented during each sampling interval.

Magnetic tape recordings were made of each type of sound to be averaged. The hundred recordings required of each particular sound were taken from exactly the same position on the chest. To keep all waveforms of each type as closely identical as possible, the volume of each breath was restricted to the minimum, and an endeavour was made to keep all percussion blows consistent.

As with chart recording, the 'advance' signal from the tape recorder triggered the analyser just before the arrival of the percussion waveform. To eliminate any possibility of drift in the delay of the trigger unit and hence to ensure that the waveforms were summed coherently, the delay was monitored by the 1 kHz generator gated via the AND gate to the counter. Any small drift was manually corrected.

For greater flexibility, the sampling rate was controlled by a square wave generator external to the analyser.

When the quiet room noise was averaged, the averager was manually triggered.

4.3.3 Results

4.3.3.1 Main Waveform - first 20 ms

A single sample scan of a 'resonant' sound from the fourth

Figure 4.10 'RESONANT' SOUND - BEFORE AVERAGING

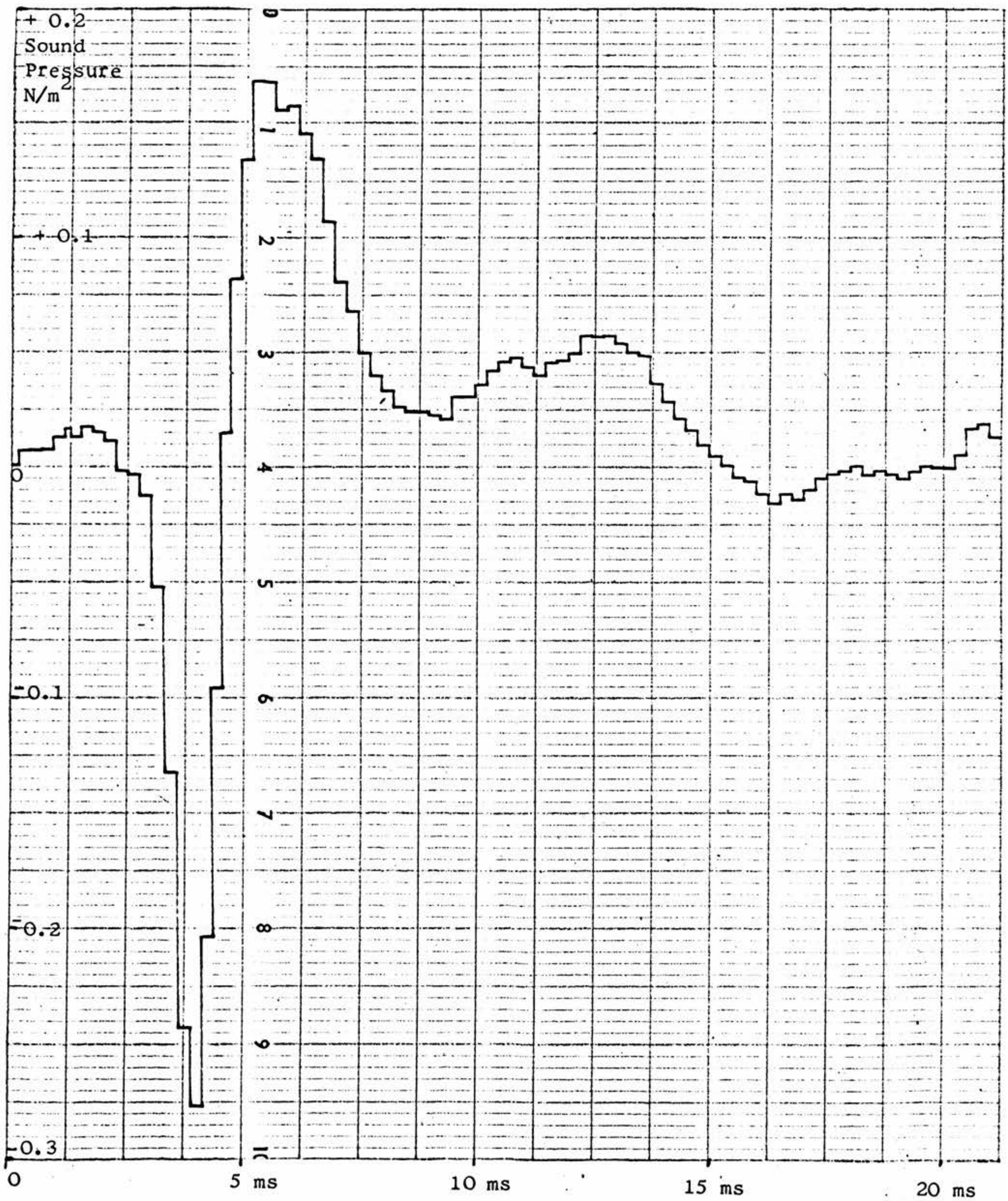


Figure 4.11 'RESONANT' SOUND - 10 AVERAGES

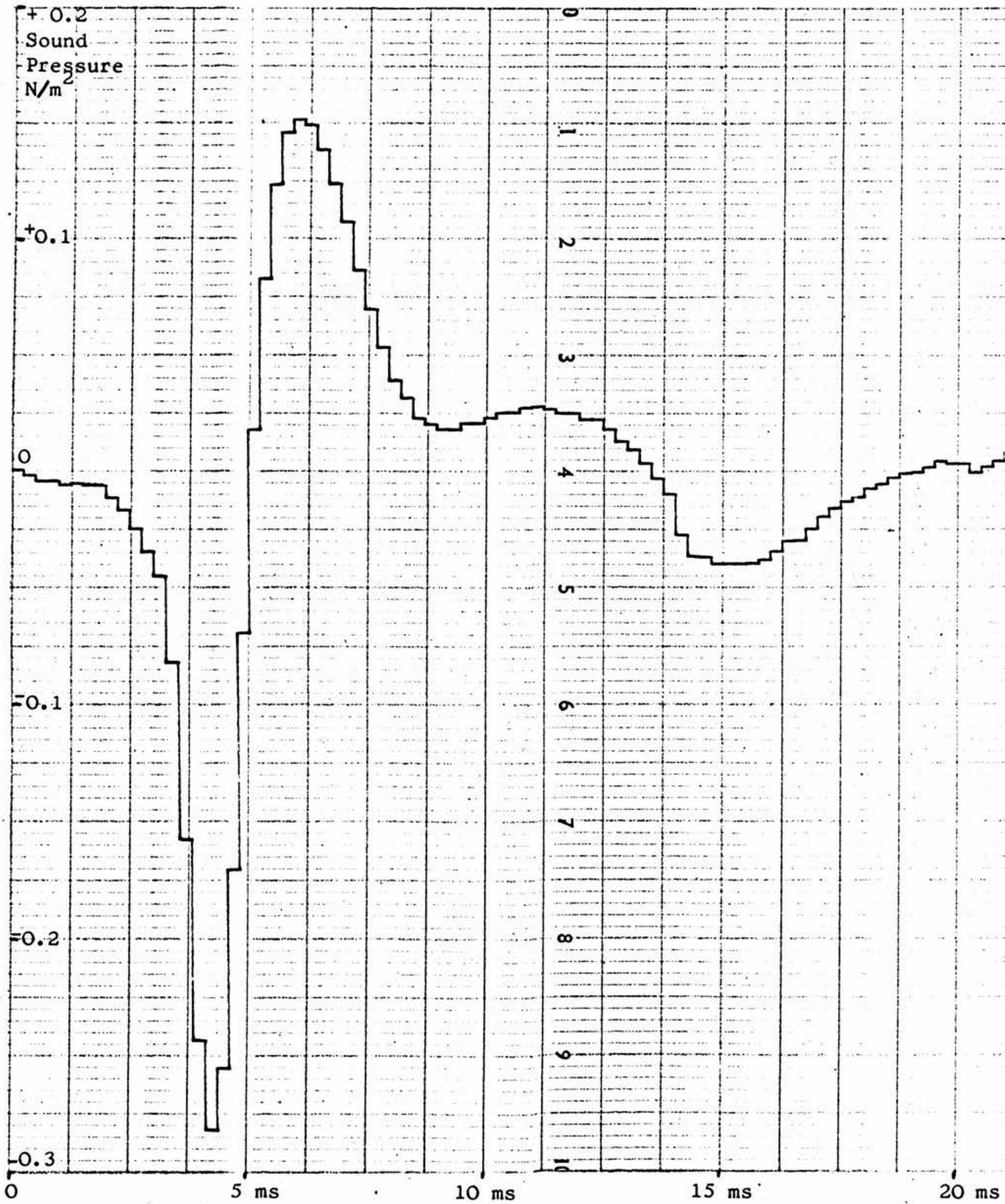


Figure 4.12 'RESONANT' SOUND - 100 AVERAGES

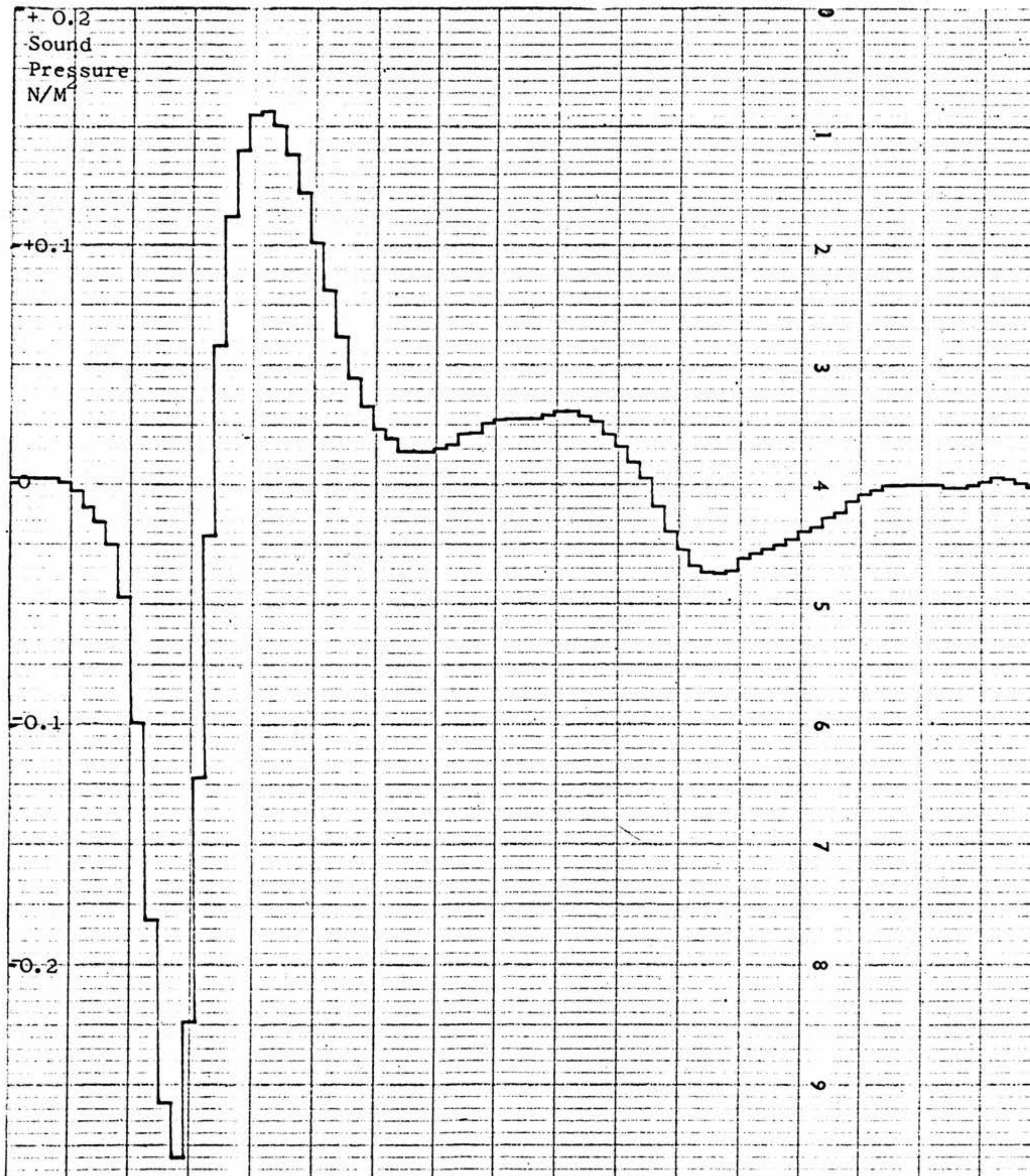


Figure 4.13 'DULL' SOUND - 100 AVERAGES

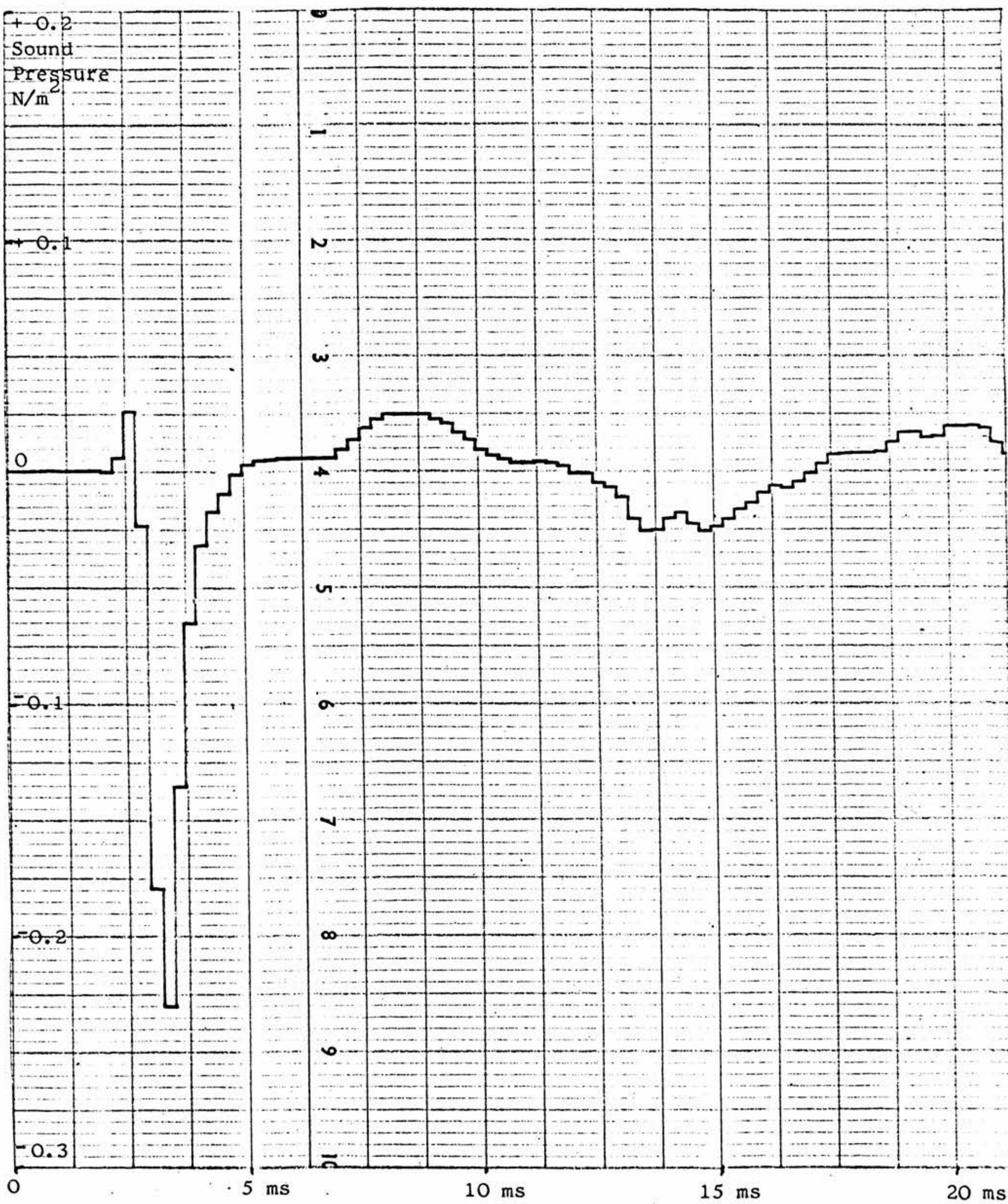
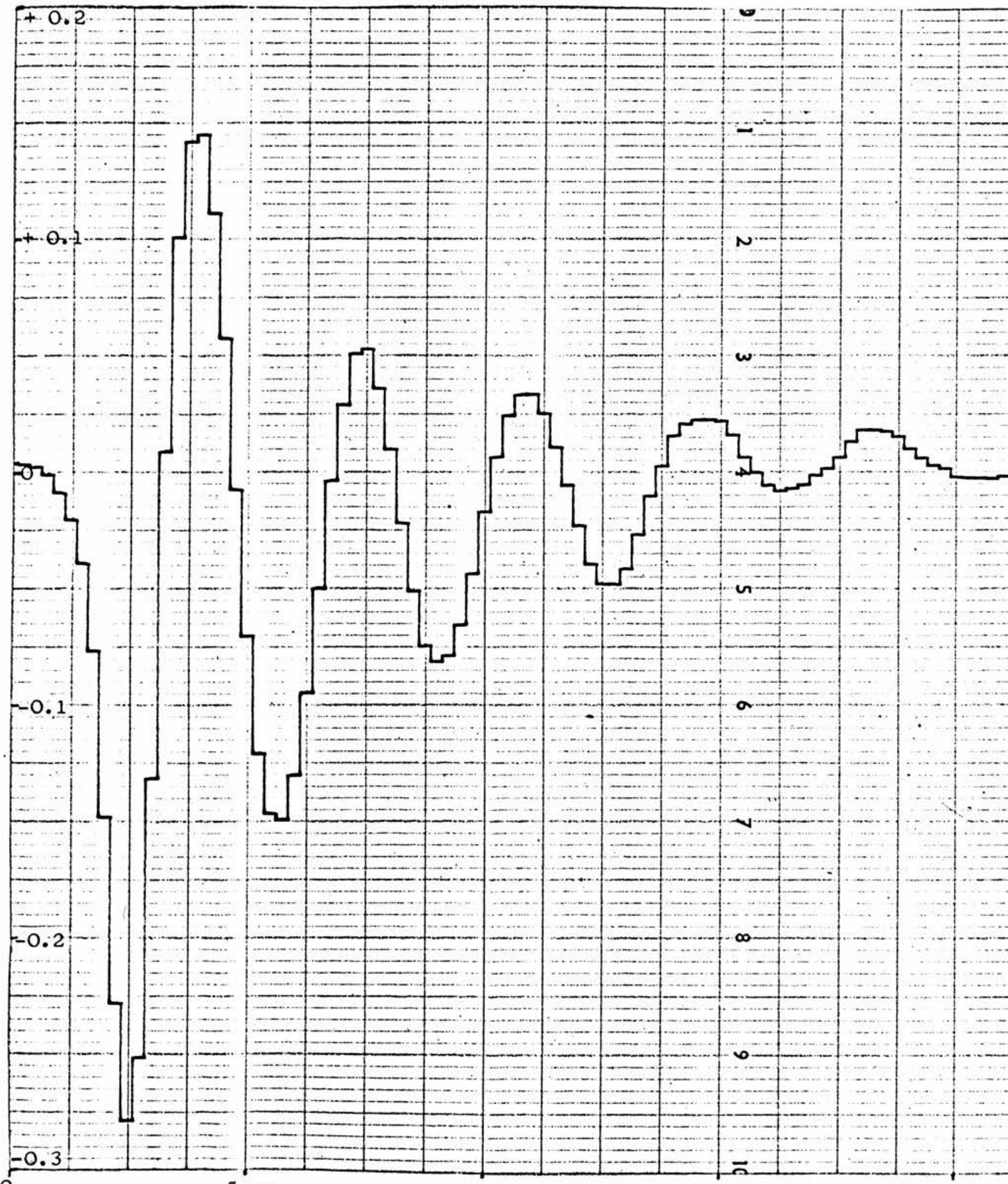


Figure 4.14 'TYMPANIC' SOUND - 100 AVERAGES



intercostal space resulted in the noisy trace of figure 4.10. Averaging ten sounds together (figure 4.11) and then one hundred (figure 4.12) produced progressively smoother curves. All three curves show the same pressure scale as the noise traces of figure 4.8. This allowed a comparison of the noise improvement to be made. The reduction in noise is partâcularly evident at the beginning and end of each waveform.

Evidence for a consistent sound pressure waveform throughout the one hundred sounds averaged, lies in the similarity of shape and size of the sampled and averaged traces. If the consistency had been poor, the averaged waveshapes would have tended to flatten out, since each section of the waveshape would not have added coherently every time.

The smoothness of the averaged curves indicated that no significant features had emerged from the noise: at least no feature whose outline was not already visible on those recordings where averaging had not been employed.

Averaging was repeated for 'dull' (figure 4.13) and 'tympanic' (figure 4.14) sounds. Again this resulted in much clearer waveshapes than obtained with direct chart recording. This was especially noticeable in the 'tympanic' sound where the fifth compression peak was still clearly visible. Nevertheless, as with the 'resonant' sound, no additional features had been discovered.

4.3.3.2 Waveform 'Tail' - second 20 ms

Only the first 20 ms of each waveshape have so far been dealt with; but what of the remaining 'tail'?

Figure 4.15 TAIL OF 'TYMPANIC' SOUND - 100 AVERAGES

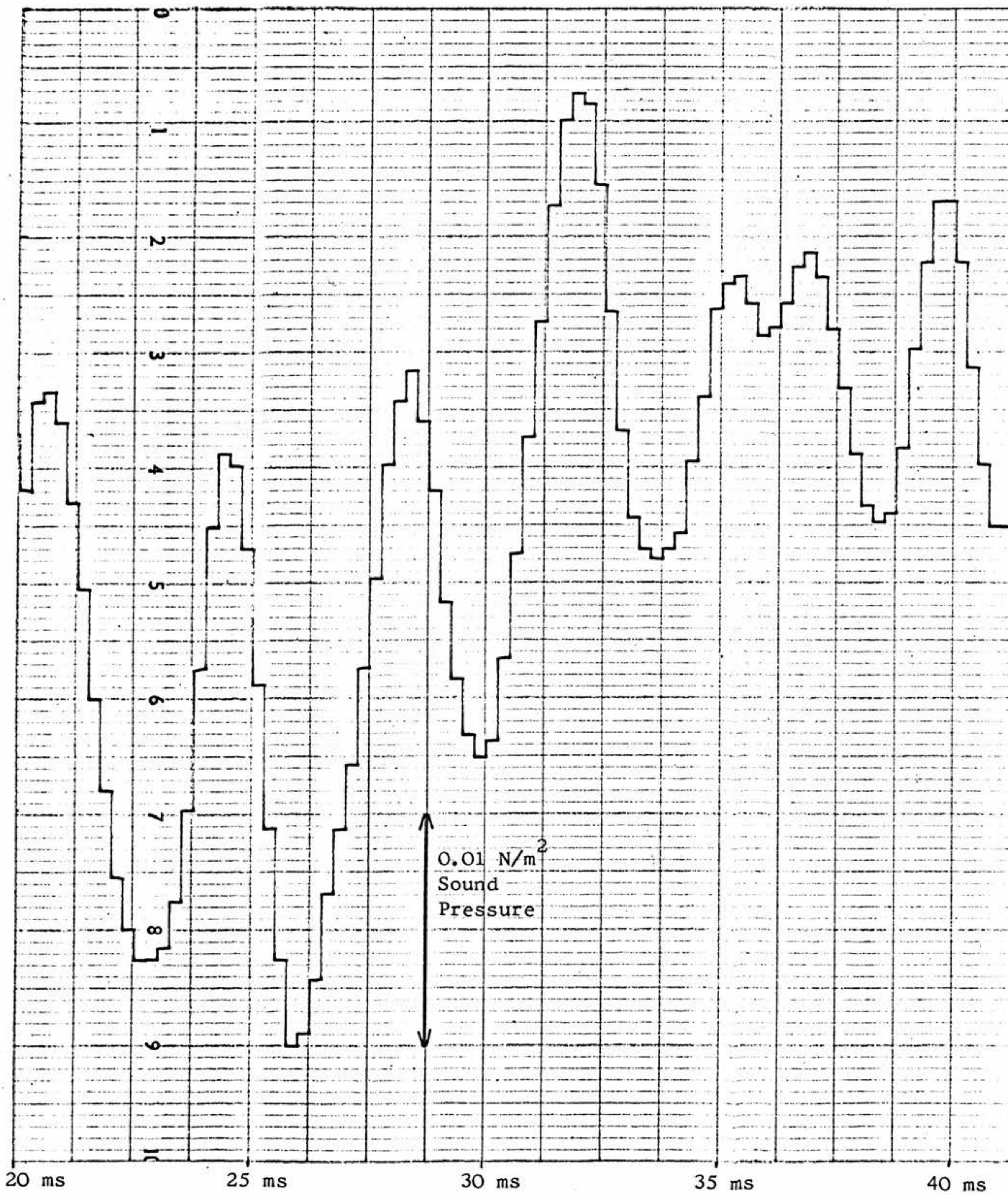
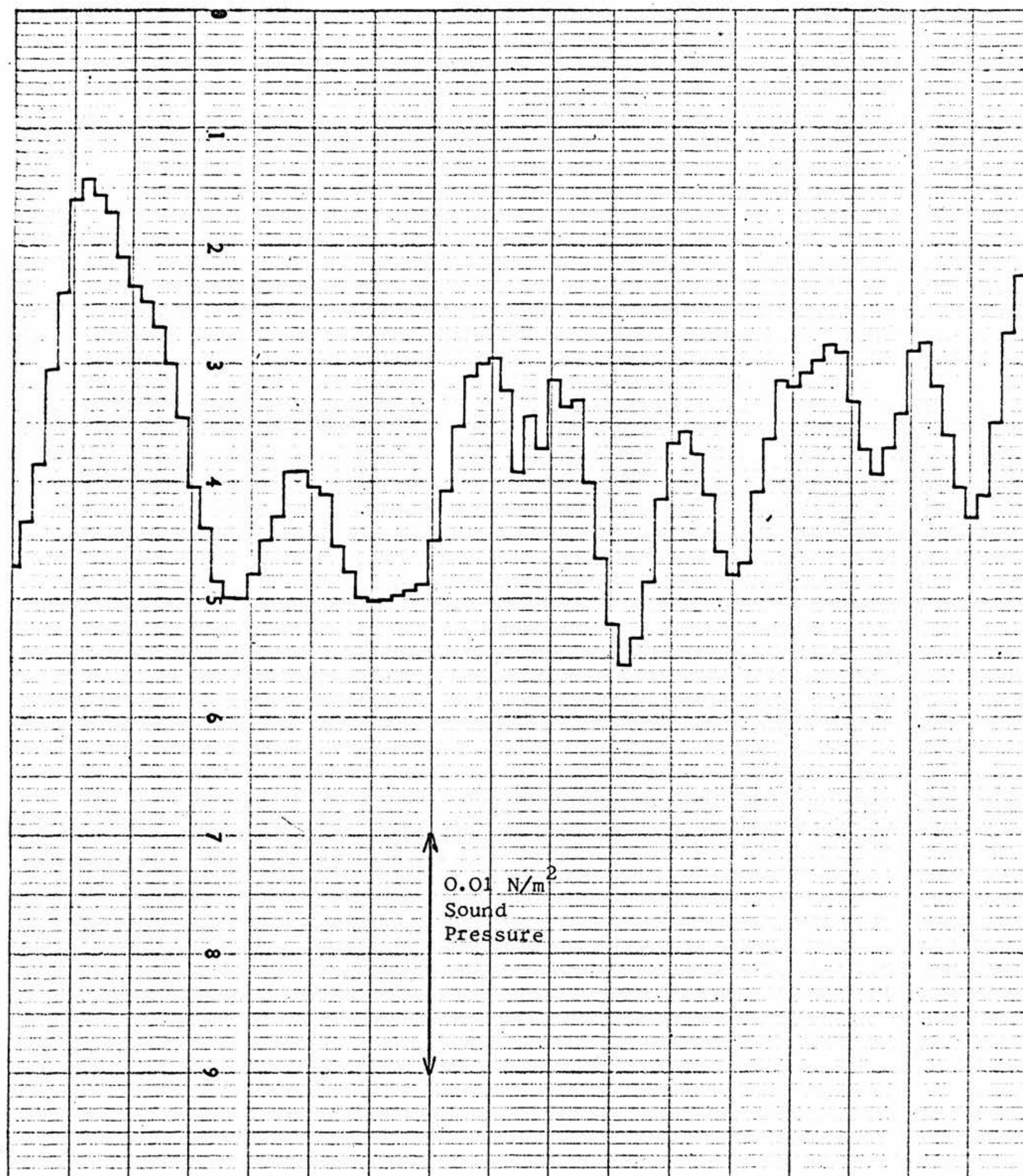


Figure 4.16 REFLECTIONS ON TAIL OF 'RESONANT' SOUND - 100 AVERAGES



Averages of the second 20 ms were next taken, firstly from the 'tympanic' sound (figure 4.15) and then from the 'resonant' sound (figure 4.16). Both traces clearly revealed characteristics not visible on the chart recordings. Note that, compared with the averages of the main waveform, the pressure scale has been expanded ten times to allow the small fluctuations to be observed.

Although the resulting erratic sound pressure on the 'resonant' trace lay well above that expected from electrical or background noise, the traces were checked for noise by averaging the first fifty and the last fifty of the 'resonant' sounds separately. Both resulting traces were almost identical to that of figure 4.16, proving that the 'tail' was undoubtedly correlated with the actual 'resonant' sound. Various recording positions in the room, however, resulted in different waveshapes in the 'tail', showing that although the tail was correlated it was not part of the original direct sound. The waveshape in the 'tail' was due to reflections. These reflections had arisen in spite of certain precautions taken to minimise them, such as placing sound absorbent material on the floor and taking the recordings at a position in the laboratory which was furthest from all reflecting surfaces.

Without the use of an anechoic chamber, reflections made it impossible to detect what waveshape, if any, existed 20 ms after the start of the 'resonant' sound.

On the 'tympanic' sound, the reflections merged with the original sound making it impossible to separate the original from the reflections on the 'tail' of the waveform.

As an anechoic chamber was not available, it was decided not to pursue the investigation into the 'tail', which under normal circumstances would, in any case, be masked in sound reflections and noise.

4.4 FREQUENCY ANALYSIS

4.4.1 Fourier Analysis

Confident that no important features of the percussion sounds had been neglected, the analysis was extended from the purely observational stage.

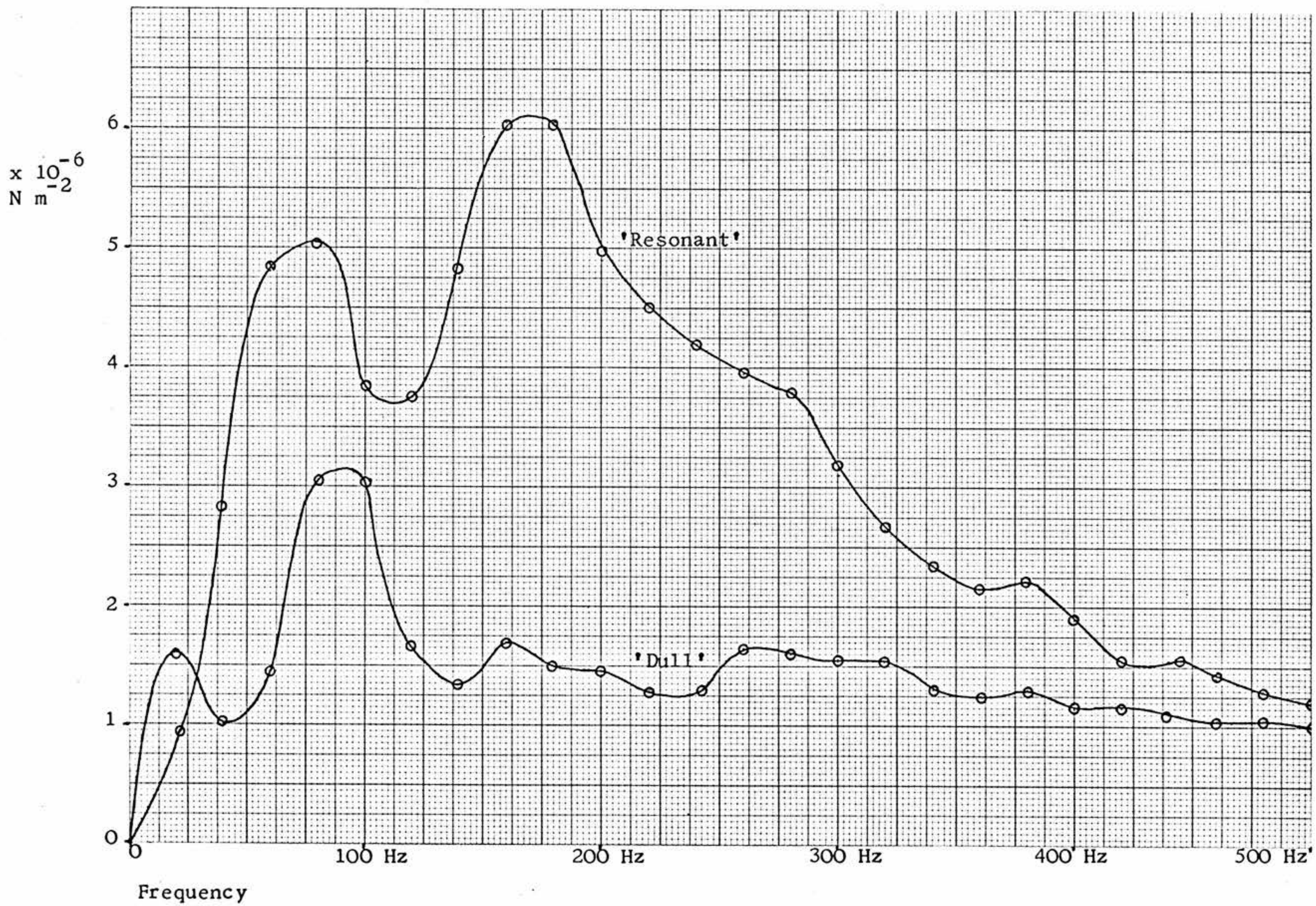
Frequency analysis of the percussion sounds was employed because it was well known that the ear was sensitive to the different frequency components of sounds. In addition, the ear is known to do a type of Fourier analysis. Littler (1965), who discusses the various theories of hearing, summarises the position thus - "the subjective phenomena of hearing .... show that the cochlea, with its neural connections to the cortex, does carry out some forms of analysis very like the separation of Fourier components on the resonance theory to lead to what may be called a pattern analysis."

Although the bulk of Casteleijn's (1961) work had been concerned with Fourier analysis of percussion sounds, it seemed worth repeating; primarily because he had come to the surprising conclusion that there was little difference in spectrum between 'resonant' and 'dull' sounds. In contrast, this investigation has emphasised the vastly different waveshapes of these sounds.

Further justification for repeating the analysis was provided by the improved signal-to-noise ratio of the averaged sounds, and the access to an IBM 360/67 digital computer for accurate calculation of both the magnitude spectrum and, for the first time, the phase spectrum.

The input data for the computer calculation were obtained from a print-out of the contents of the one hundred channel analyser after

Figure 4.17 FOURIER MAGNITUDE SPECTRUM - 'RESONANT' AND 'DULL' SOUNDS



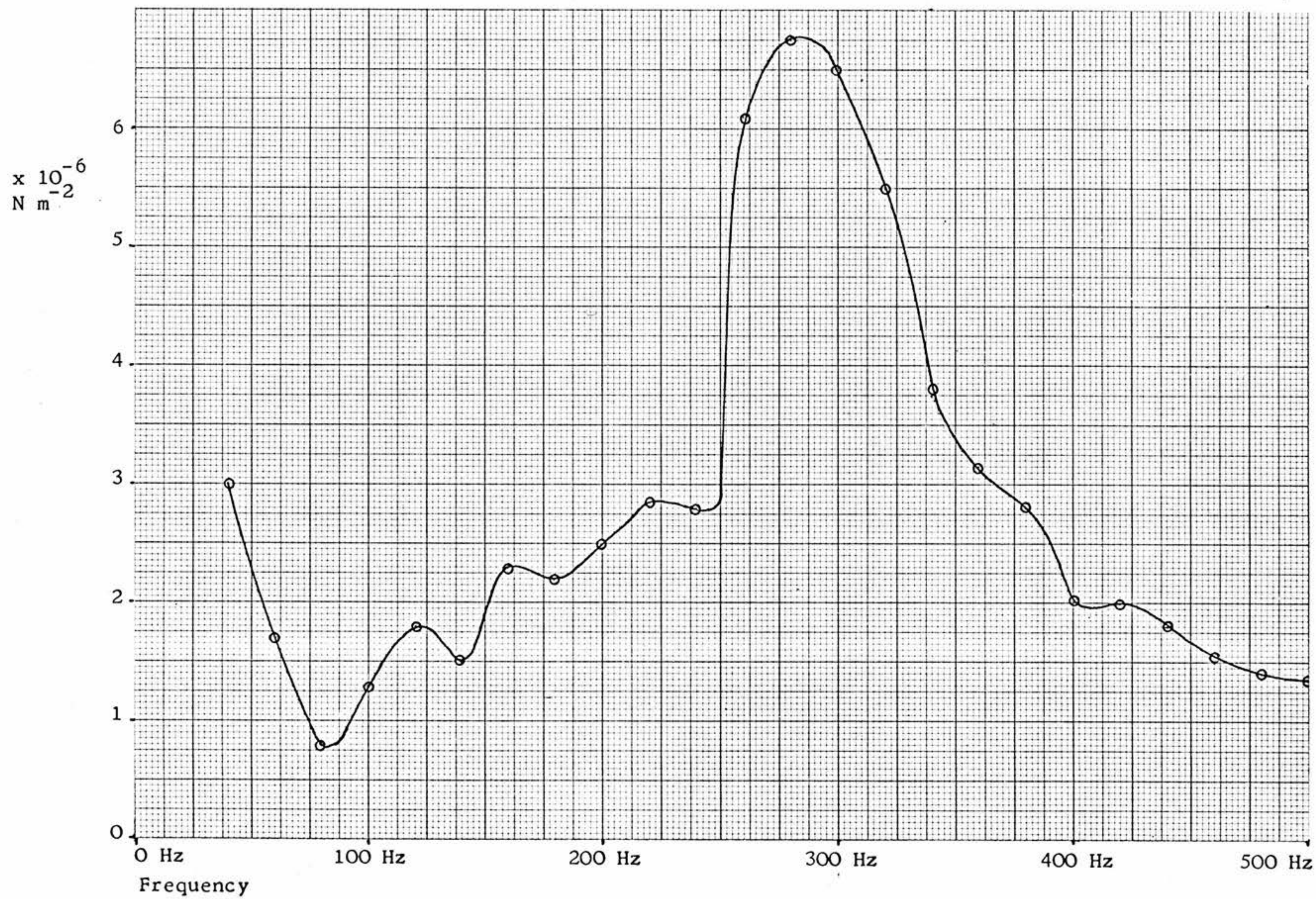
averaging. Written into the program was the necessary information required to convert the analyser counts into the equivalent sound pressures. Values of magnitude and phase were calculated and tabulated for all frequency components from 0 Hz, in 20 Hz steps, up to 2 kHz. However, only frequencies from 0 Hz to 500 Hz were plotted in the Fourier spectrums because the frequency components quickly decayed to insignificant values. Listed in the table below is the difference between the magnitude of the component at 500 Hz and that at 2 kHz, for the three sounds analysed.

Percussion Sound	Magnitude of 2 kHz component relative to 500 Hz component
'Resonant'	- 49 dB
'Dull'	- 40 dB
'Tympanic'	- 37 dB

4.4.2 'Resonant' and 'Dull' Sounds

Fourier analysis of the 'resonant' and 'dull' sounds of figures 4.12 and 13 yielded the magnitude spectra of figure 4.17. Immediately apparent is the large difference in frequency spectra between the two sounds. The 'resonant' sound contains a far greater proportion of its energy at the lower frequencies than the 'dull' sound. This explains why, in the subjective judgement of the physician, 'resonance' is described as a 'low frequency sound'. 'Dullness' tends to have a far greater spread in energy, and hence in comparison with the 'resonant' sound has a large proportion of its energy at the higher frequencies. Therefore to the ear, 'dullness' appears higher in frequency than 'resonance' - hence accounting for its description as a 'high frequency vibration'.

Figure 4.18 FOURIER MAGNITUDE SPECTRUM - 'TYMPANIC' SOUND



The use of the term 'vibration' is, however, misleading as no vibrations are present - a fact which is easily verified from figure 4.13. It is only the frequency spectrum of the short impulsive sound which leaves the impression of a high frequency sound.

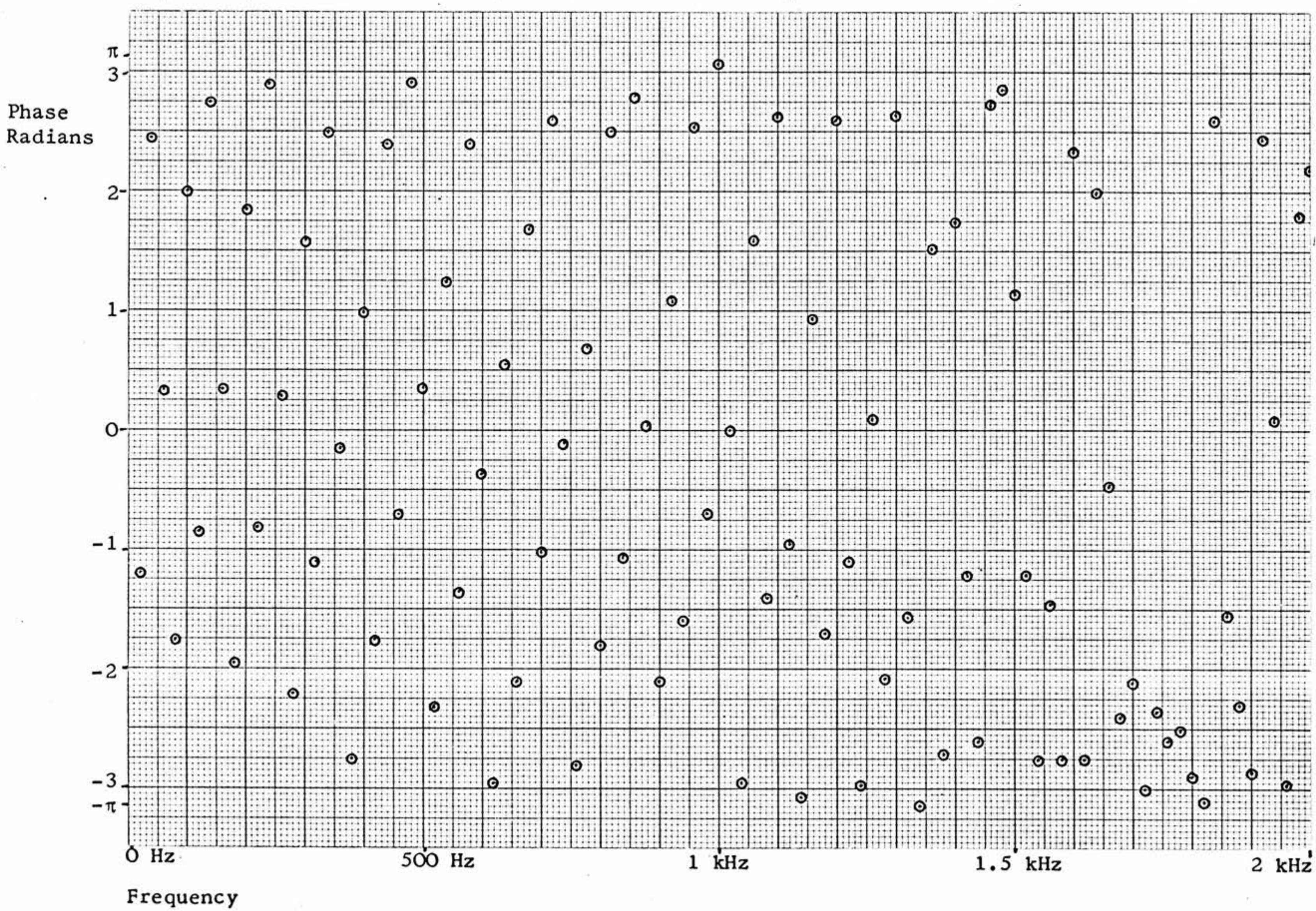
Previous attempts at determining a 'fundamental frequency' (see section 1.5.2.) have not been aimed as was thought at finding the frequency of some periodic variation contained in the percussion sounds. At this stage, all that can be concluded from the results these investigators obtained is that their measure of 'frequency' was one which was weighted in some way by the distribution of energy throughout the spectra of the sounds they attempted to analyse.

4.4.3 'Tympanic Sounds

Fourier analysis of the 'tympanic' sound of figure 4.14 yielded the spectrum of figure 4.18. In comparison with the spectra of 'resonant' and 'dull' sounds, a much more distinctive peak was obtained. Care must always be taken when speaking of 'frequency', because as seen above, no specific frequency could be noted in the time domain recordings of the waveforms of either the 'resonant' or 'dull' sounds.

Consider for the moment, two simple cases of Fourier transforms. A continuous sinusoidal variation transforms into a single line spectrum, and this spectrum could without doubt be described as containing a single frequency. The second case, which this time approaches the spectrum in question, is the transform of a damped sinusoid such as $\sin \omega t \; e^{-t/\tau}$. This spectrum has a smooth symmetrical curve with a single peak at ω. The breadth of the peak increases with damping, which results from a decrease in τ. In such a case it is permissible to conclude that the frequency ω exists in the time domain waveshape, although this does

Figure 4.19 FOURIER PHASE SPECTRUM - 'RESONANT' SOUND



not hold for those frequencies on either side of the peak at ω, even although they too are contained within the spectrum.

Returning to the spectrum of figure 4.18 it can be seen that although there is a distinctive peak at 280 Hz, the spectrum is by no means smooth or symmetrical. This is to be expected since the waveform of the 'tympanic' sound (figure 4.14) is not a perfect exponentially damped sinusoid. Nevertheless, if the average period is measured from the waveform of figure 4.14, a time of approximately 3.5 ms is obtained. When converted to a frequency, a value results which is approximately equal to 280 Hz - the frequency previously measured at the peak of the spectrum.

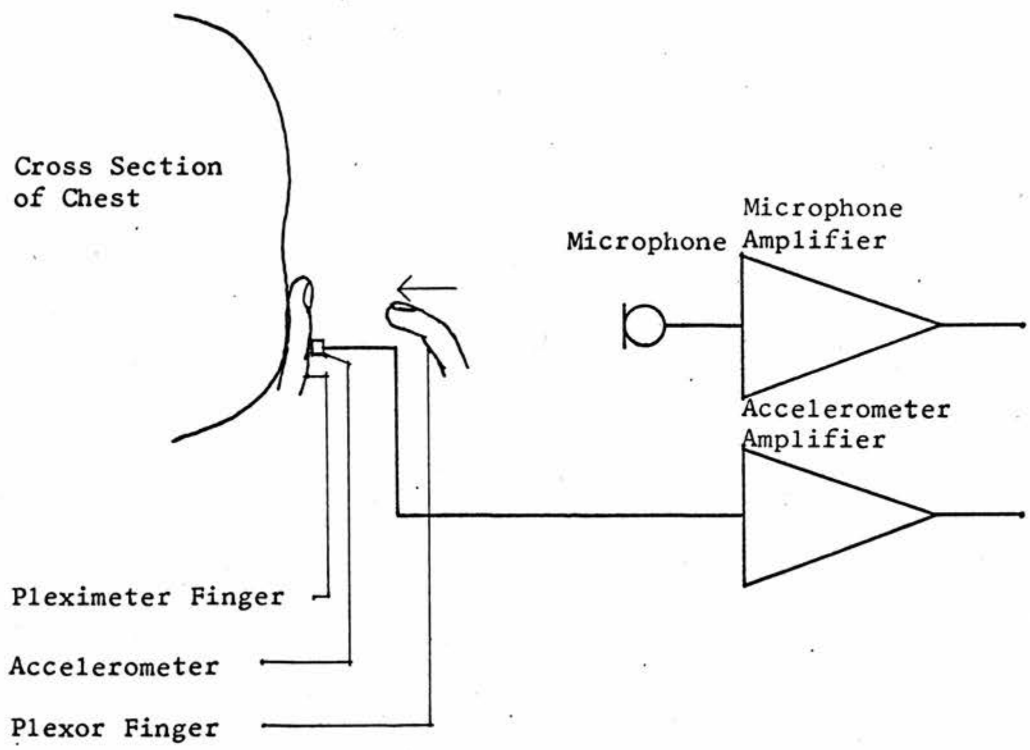
Hence there is some justification for describing 'tympanic' sounds in terms of a particular frequency, even although no such justification exists in the interpretation of 'resonant' or 'dull' sounds.

4.4.4 Phase Spectra

When the magnitude spectrum is obtained from the time domain waveform, some information carried by the waveform is lost. This information is contained in the phase spectrum. Without both the magnitude and phase spectra, the original time domain waveform could not be reconstructed. The concept of phase information is difficult to interpret other than in the reconstruction of the original waveshape, where the phase of all frequency components is adjusted at $t = 0$ on the time domain.

Along with the magnitude spectra of the 'resonant', 'dull' and 'tympanic' sounds, the phase spectra were also calculated. The phase spectrum of the 'resonant' sound is shown in figure 4.19 to illustrate the typical spectrum obtained in all three cases. The points plotted

Figure 4.20 SOUND PRESSURE AND PLEXIMETER ACCELERATION MEASUREMENT SYSTEM



were for all those frequencies for which the magnitude spectrum had been calculated. As continuous spectra are produced for transient sounds, the points should have been joined in a way similar to figures 4.16 and 17. However, due to the oscillatory nature of the phase spectra, it was impossible to interpolate between points.

As the phase spectra did not aid the understanding of percussion sounds, further consideration of them was abandoned.

4.5 PHYSICAL EXPLANATION OF WAVESHAPE AND FORMULATION OF PHYSICAL MODEL

The analysis then returned to the time domain for a more detailed study of the waveform.

Observations made on the wavetraces were correlated with the chest motion which followed a percussion blow. From the observations, a theory of the production of the general waveshape was produced and a physical model formulated.

In addition, the beginning of the rarefaction was studied in some detail and four of its features isolated.

4.5.1 Acceleration Measurements

Some of the observations of this section required the measurement of the pleximeter acceleration. This was achieved with the help of a small accelerometer (D.J.B. Type A/04) attached to the pleximeter finger (figure 4.20).

An accelerometer, it must be emphasised, measures only its own acceleration. However, as it was firmly bound to the pleximeter finger, and as its mass (0.1 oz) was insignificant compared with that of the finger, it measured the acceleration of that section of the finger to which it was attached. But as the finger was non-rigid, its motion could not be described by that of one point. Nevertheless, as the finger was firmed in a straight position and as the direction of the percussion blow lay parallel to the principal axis of the accelerometer, the acceleration measured was taken as a good indication of the acceleration of the pleximeter. In any case the accuracy was sufficient for a purely qualitative study.

To allow the sound pressure and the pleximeter acceleration waveforms to be compared, they were recorded together on the two channel chart recorder. Before a final comparison was made, the recordings had to be corrected for the propagation delay of the sound. Knowing the following,

Microphone-to-chest distance	=	15 cm
Sound propagation velocity	=	344 m/s

a propagation delay of 0.44 ms was to be expected. With a real-time chart recording scale of 1.7 ms/cm, this necessitated displacing the sound waveform 2.5 mm to the right.

4.5.2. Features on the Rarefaction Waveform

Before consideration is given to the means by which the whole waveform is produced, four different features of the rarefaction pressure spike which have been isolated will be discussed below. The third and fourth features to be considered are extremely important since they provide an understanding about some quite noticeable components of the waveshape. The first two are included to complete the study of this portion of the waveform.

4.5.2.1 Initial Pressure Rise

Even before the plexor finger touches the pleximeter, a pressure rise can be detected as the air trapped between the two fingers is

Figure 4.21 ORIGIN OF INITIAL PRESSURE RISE

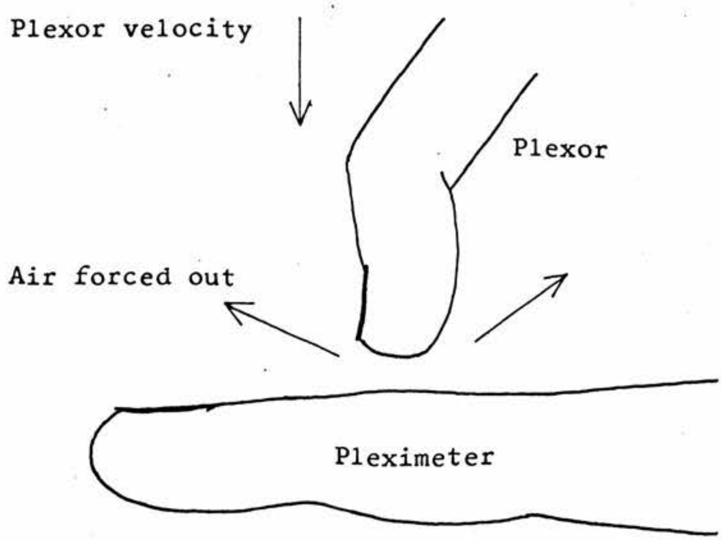
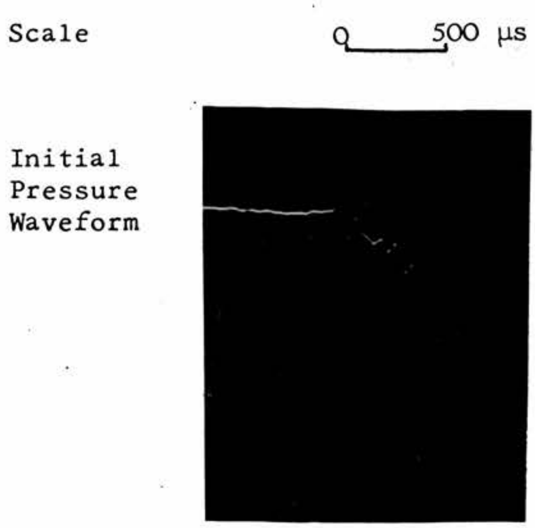


Figure 4.22 INITIAL PRESSURE RISE FOLLOWED BY SKIN CONTACT SOUND



forced out of the quickly diminishing gap just before contact is made. Figure 4.21 illustrates this, and the resulting small pressure rise can be observed on the trace of figure 4.22. This waveform was photographed from an oscilloscope trace, the microphone being connected directly to the oscilloscope for an increased high frequency response.

4.5.2.2 Skin/skin Contact

Contact, when it does occur, forces the pleximeter finger and the chest tissues surrounding it to move, creating a fall in pressure. (See figure 4.22). This greatly amplified pressure waveform also reveals a high frequency oscillation, greater than 10 kHz in this case. The oscillation appeared to be the result of frictional rubbing between the two skin surfaces as the fingers make contact.

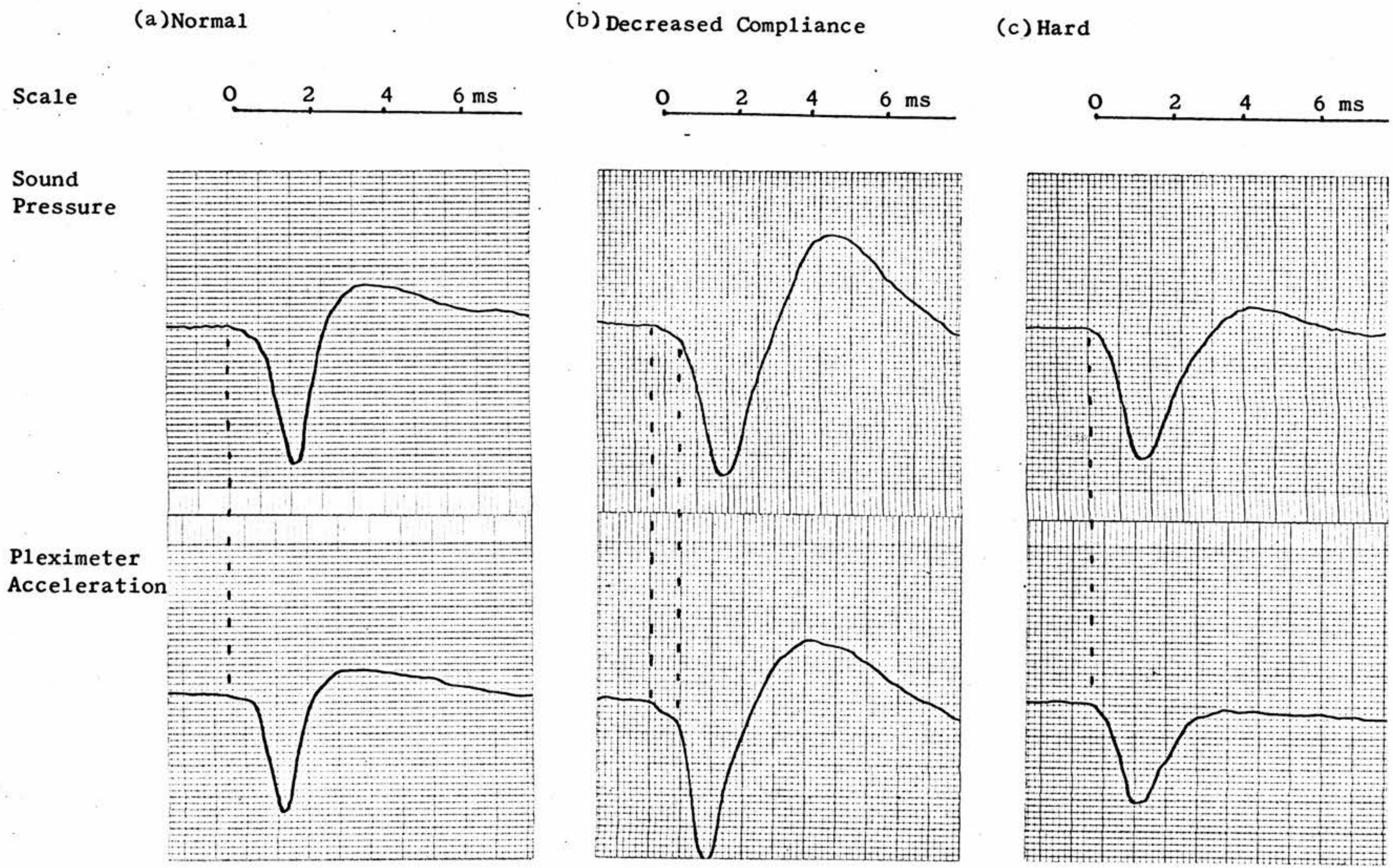
No two traces produced by skin contact were identical since they were affected by such factors as position, velocity and angle of impact. Figure 4.22 can therefore be considered as being only one particular example. Nevertheless it is illustrative of the type of sound trace produced by contact of the two skin surfaces.

It should be noted that this contact oscillation and the preceding pressure rise were not of sufficient importance to warrant detection by the normal percussion recording system, but are included here for the sake of completeness.

4.5.2.3 Plexor Tip Firming

The tip of the plexor finger is soft and fleshy, and so on impact it must first be firmed before the main kinetic energy of the percussion blow can be transferred to the chest.

Figure 4.23 EFFECT OF PLEXOR TIP



Observations on the effect of this tip were made by comparing recordings of the sound pressure and pleximeter acceleration, the sound propagation delay having been compensated for on all recordings. (Unless otherwise stated the full scale deflection of all pleximeter acceleration traces will be approximately $\pm$ 40 ms^{-2}.) A heavy pleximeter force was used to minimise the effect of chest tissue firming (see section 4.5.2.4).

The influence of the plexor tip on the sound pressure is best explained by studying a few examples. Three different types of tip were employed in producing the traces of figure 4.23.

With a normal plexor tip there is a time lag after contact before the large sound pressure and acceleration changes begin to be recorded. The motion of the pleximeter is only slight as the plexor tip is firming, but thereafter the pleximeter is set properly in motion and the large sound pressure rarefaction generated.

Using a firmer tip (with decreased compliance) produces, as expected, a greater initial acceleration (figure 4.23b). To obtain this tip, the top of a normal plexor was tightly bound while leaving the tip uncovered.

Finally, trace (c) shows the effect of a hard (Perspex) plexor tip. No separate response from the plexor tip can be identified, as it could be in the two previous cases.

4.5.2.4 Chest Tissue Firming

The pleximeter finger, in contact with the chest, rests on a layer of soft tissue which must be compressed before the percussion blow strikes the finger. Otherwise some of the energy of the blow would be needlessly expended in firming the tissues.

Figure 4.24 LIGHT PLEXIMETER FORCE

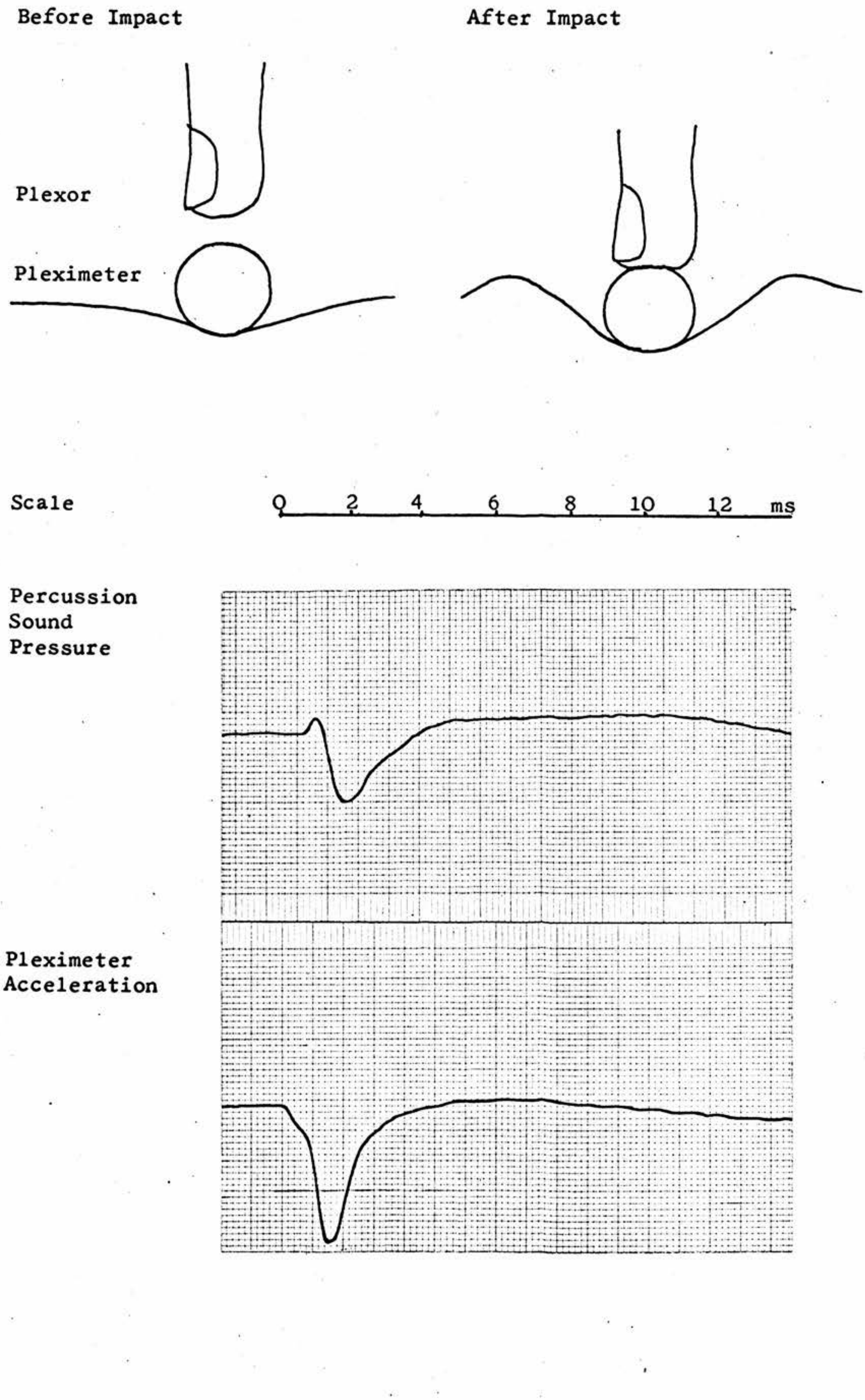
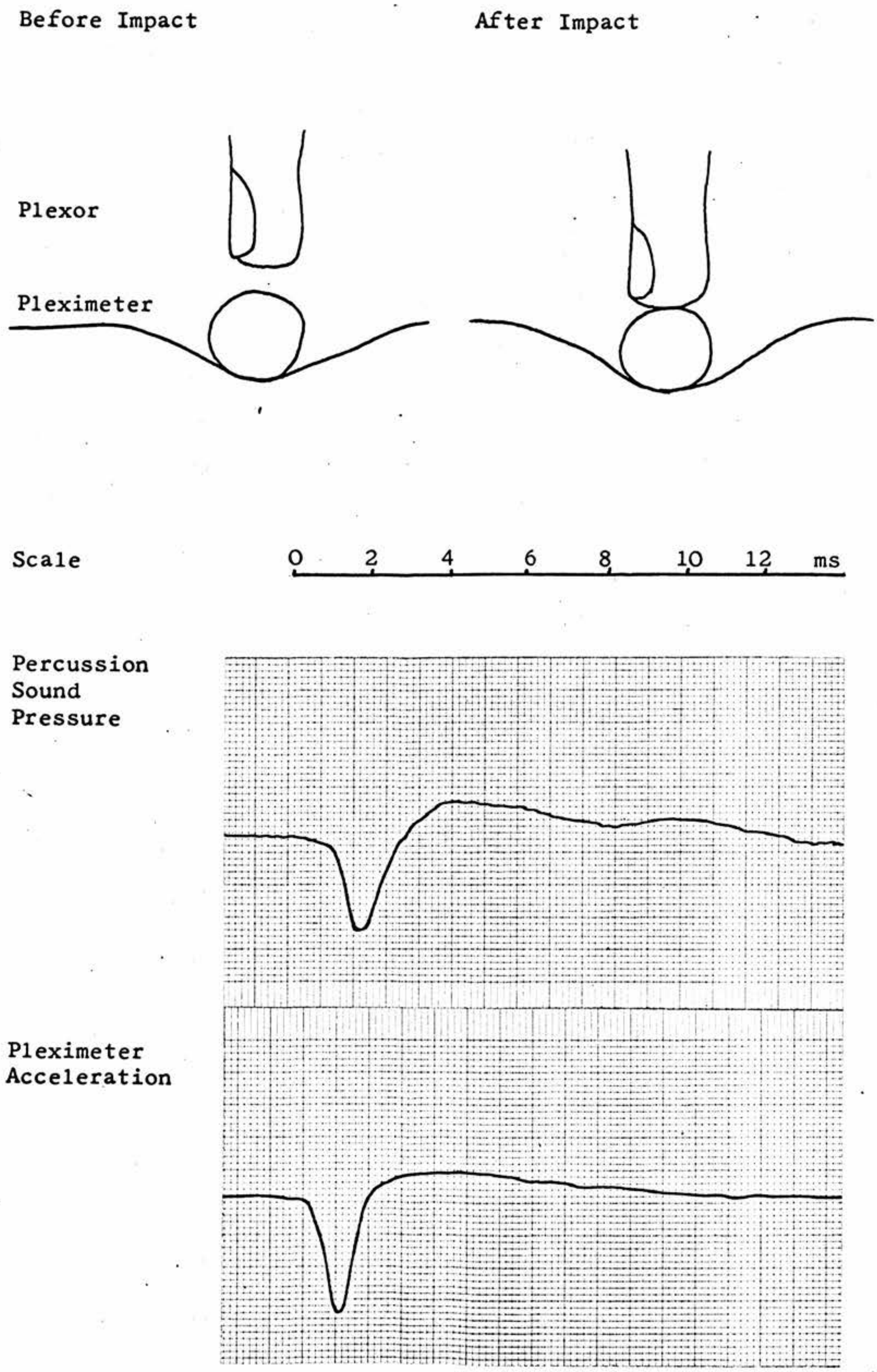


Figure 4.25 NORMAL PLEXIMETER FORCE



In considering the effect of the chest tissues, the influence of light, normal and heavy pleximeter forces on the pressure waveshape were examined.

Virtually no tissue firming is afforded by a light pleximeter force (figure 4.24) and so the first result of the percussion impact is to throw up tissue on either side of the pleximeter finger, creating an increase in pressure. Over the flabby tissue of an obese subject this effect is particularly difficult to avoid. Hence when percussion sounds were to be analysed for this thesis, obese people were deliberately avoided so as not to add an extra and unimportant feature to the waveform.

An intermediate force (figure 4.25) prevents this outward motion of tissue. But as the tissues are not fully compressed, the percussion blow first acts predominately on an area in its immediate vicinity and only as the pleximeter is depressed further does a greater area of tissue begin to move. Only then is a large pressure decrease noted.

A heavy pleximeter force allows the pleximeter motion to be quickly coupled to the chest rib cage (figure 4.26).

It must be noted that the influence of the plexor tip and chest tissues cannot be completely separated. For although the effect of plexor tip firming predominates first and the chest tissues next, the initial pressure decrease is due to both effects acting simultaneously.

4.5.3 General Waveform

Consideration will now be given to those features which make the greatest contribution to the whole waveform.

As will already have been observed, a marked similarity exists between part of the sound pressure waveshape and the pleximeter acceleration. For example, see the 'resonant' waveshape of figure

Figure 4.26 HEAVY PLEXIMETER FORCE

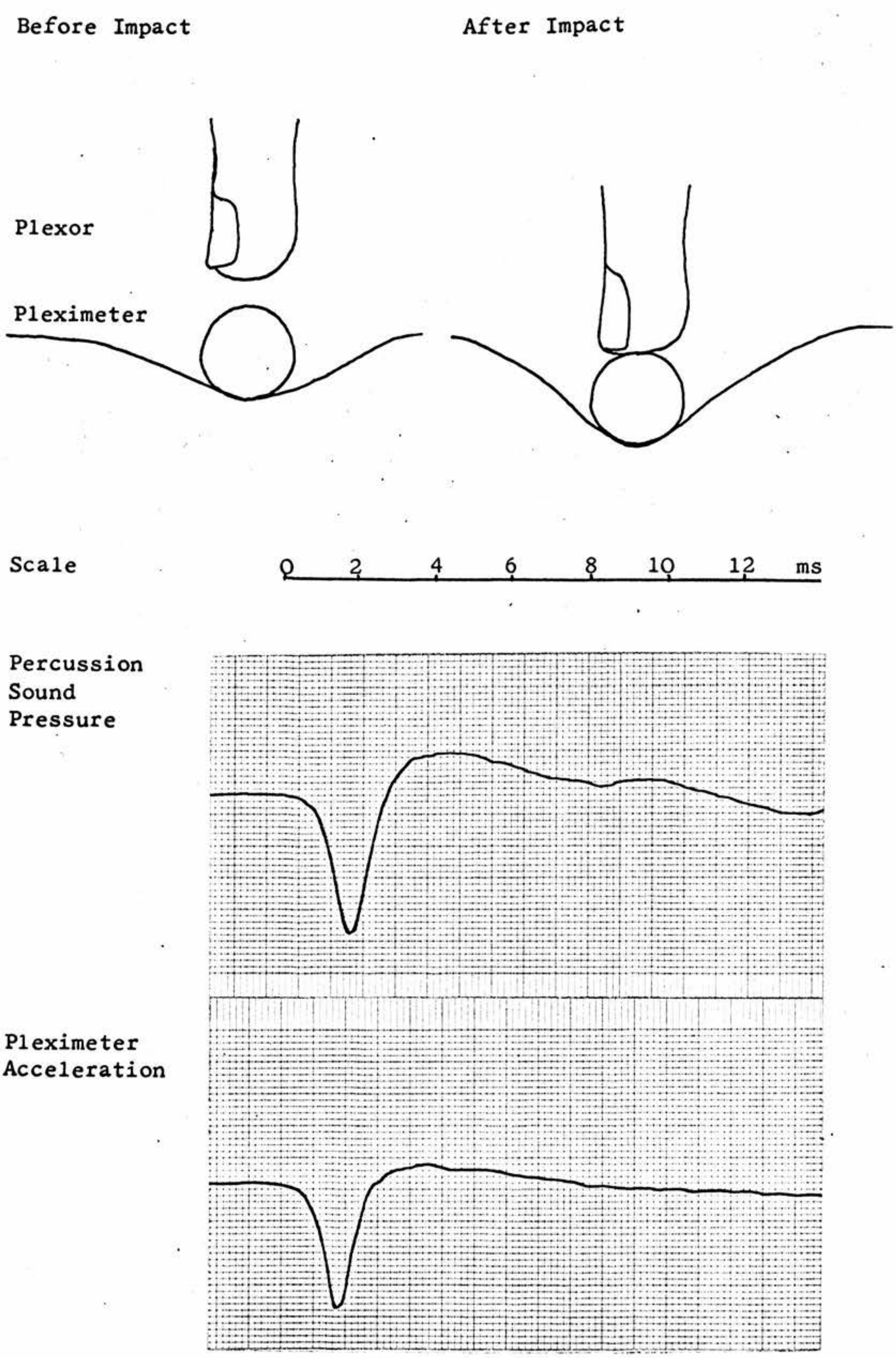
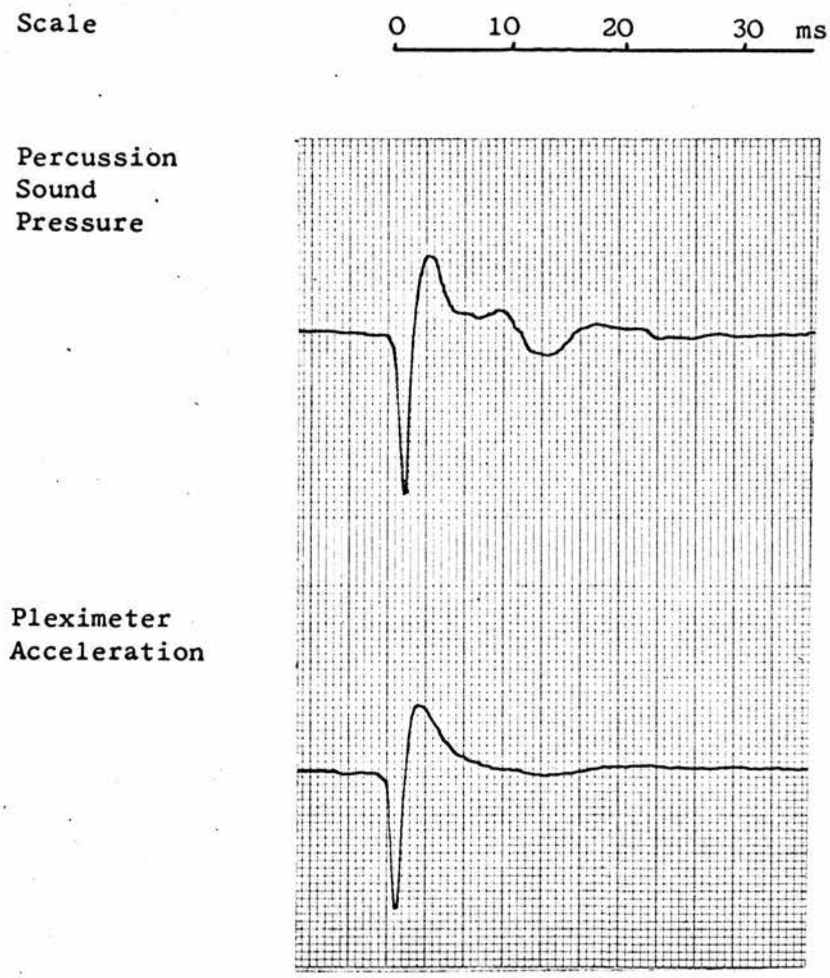


Figure 4.27 PLEXIMETER ACCELERATION



4.27. An explanation of this observation can be obtained using a simple or point source model.

First, represent the sound source by a sphere. Since the sound is radiated outwards from an area of the chest the wavefront will tend to be spherical.

The relationship between the sound pressure (p) radiated by a pulsating sphere, and the particle velocity (u) can be expressed (appendix A1) by

$$cp + r\frac{\delta p}{\delta t} = \rho_o cr \frac{\delta u}{\delta t} \qquad \text{A1 - 7}$$

where c = velocity of propagation

r = radius of spherical wave

ρ_o = equilibrium density of air

At the surface of the sphere, the air particle velocity equals the surface velocity. Formula A1-7 can be simplified if the minimum wavelength is much greater than the radius of the sphere. If the maximum frequency component of percussion sounds is again taken as 1.5 kHz, the minimum wavelength is

$$f_{min} = \frac{c}{f_{max}} \doteq 23 \text{ cm}$$

A 23 cm radius sphere would have a much greater diameter than the breadth of a normal chest, and so 23 cm is much greater than the radius of the model sphere required. Hence the above formula (A1 - 7) reduces to that of a simple source.

$$p_a = \rho_o a \frac{\delta u_a}{\delta t} \qquad \text{A1 - 8}$$

a = radius of source

Therefore the sound pressure at the surface of the model sphere is proportional to the surface acceleration. In addition, since the

Figure 4.28 MEASUREMENT OF ACCELERATION OF RIB ADJACENT TO PERCUSSION AREA

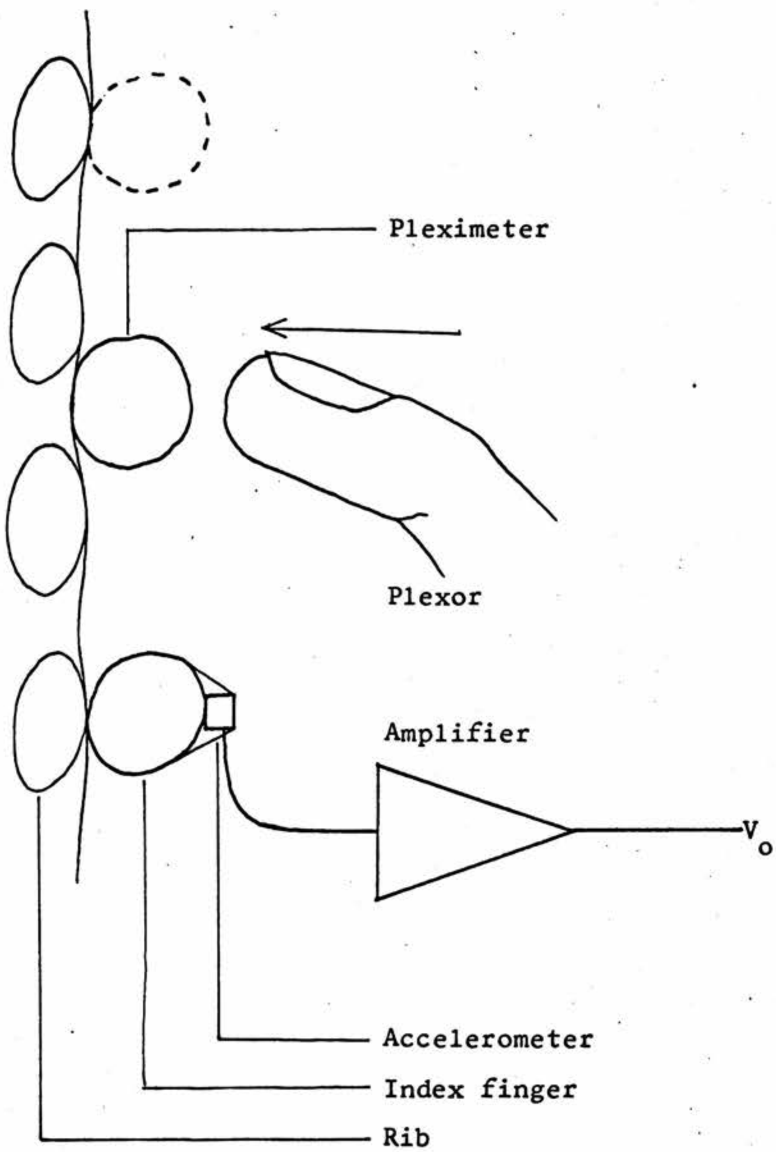
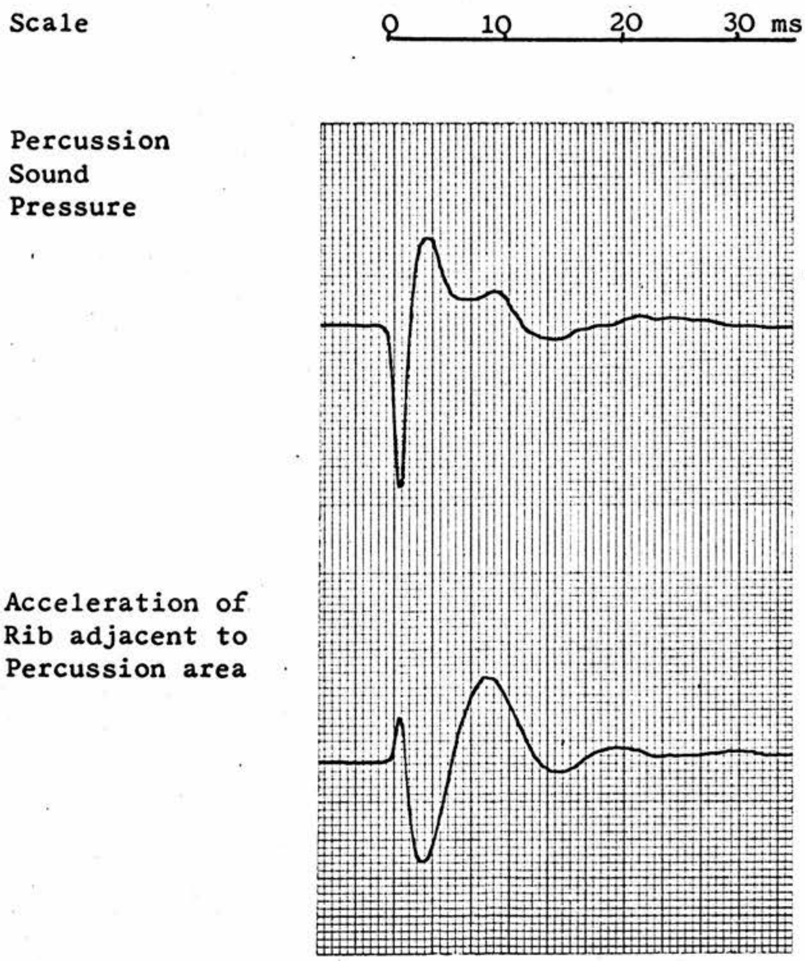


Figure 4.29 ACCELERATION OF RIB ADJACENT TO PERCUSSION AREA



sound pressure obeys the wave equation its phase does not change with propagation;

$$p = \frac{1}{r} \cdot f(ct - r) \qquad \text{A1 - 5}$$

Only a reduction in pressure amplitude and a propagation delay are introduced. Therefore if the delay is taken into account the pressure of the waveform remains directly proportional to the acceleration of the surface of the simple source.

This was a useful first approximation to the physical model being formulated as it explained the relationship between part of the sound pressure waveform and the motion of the pleximeter, and hence also the motion of the rib or ribs below it - two ribs if the pleximeter lay in an intercostal space and one if it lay along a rib.

However, as the simple model did not fully explain the sound pressure waveshape, the effect of the areas on either side of the pleximeter had to be considered; in particular the ribs above and below the percussion area. To measure the acceleration of these ribs an accelerometer was attached to one of the fingers adjacent to the pleximeter. This finger was then pressed firmly against a rib above or below the percussion area as shown in figure 4.28. Attempts had previously been made to hold the accelerometer against the rib with bandaging, but this proved to be unsatisfactory because an adequate pressure between the accelerometer and rib could not be maintained.

The relationship between the sound pressure and the acceleration of an adjacent rib can be observed in figure 4.29. (The acceleration trace has a full scale deflection of $\pm$ 4 ms^{-2}). Again it must be noted that the accelerometer measures only its own acceleration; but since the finger was pressed firmly against the rib, it followed the

Figure 4.30 SIMPLE PHYSICAL MODEL

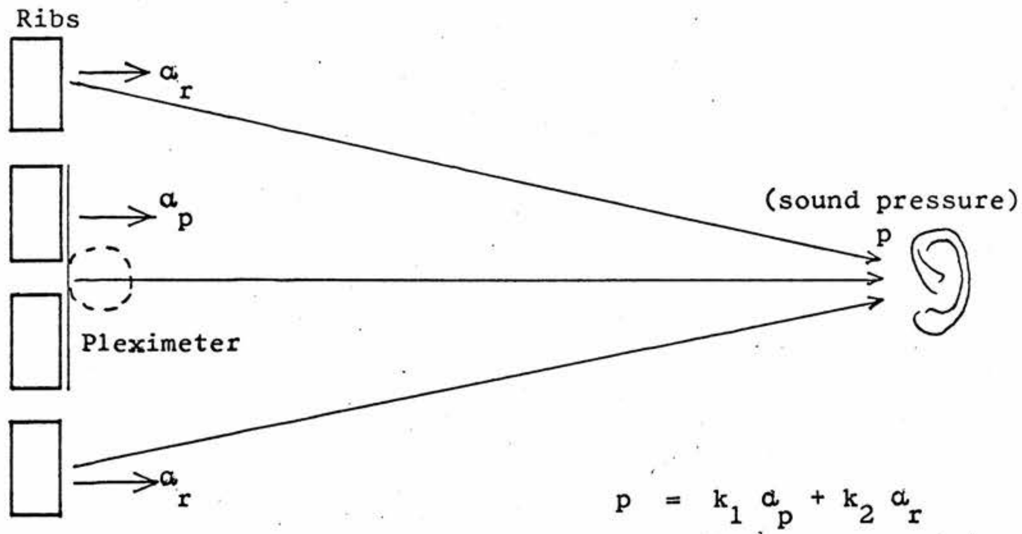
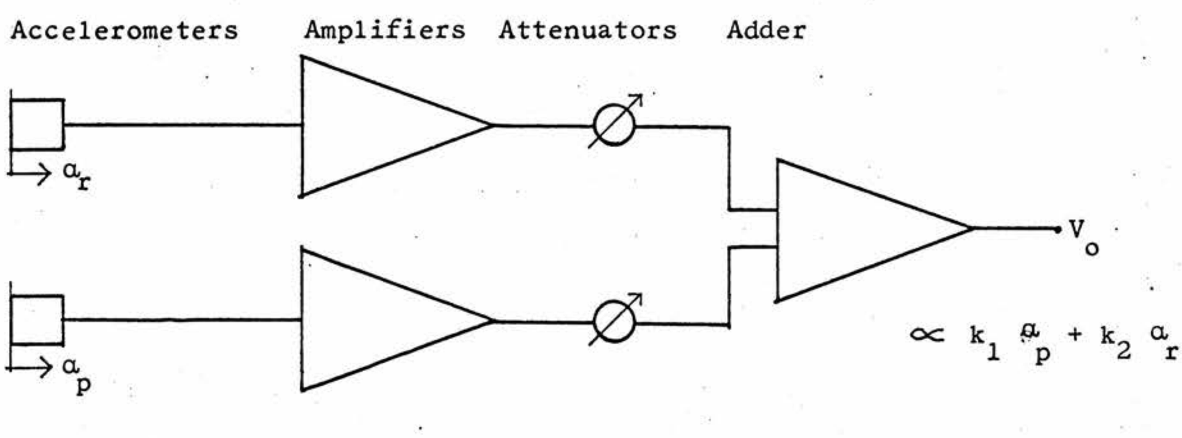


Figure 4.31 SUMMATION OF ACCELEROMETER OUTPUTS



motion of that rib closely enough for qualitative observations to be made.

From figure 4.29 it can be observed that the acceleration of the adjacent rib does in fact contribute to the sound pressure waveform - this is especially noticeable on the second compression peak.

Although the illustration is for the rib below the percussion area, a similar response can be produced from the rib on the upper side if an air cavity underlies both.

4.5.4 Physical Model

A simple model was proposed (figure 4.30) incorporating those observations.

The motion of the pleximeter (in an intercostal space) and the two ribs with which it is in contact contribute $k_1 \alpha_p$ to the sound pressure at the ear. The two other ribs adjacent to the percussion area, together add $k_2 \alpha_r$ to the sound pressure, where

p = sound pressure at a fixed distance from the model

k_1, k_2 = constants

α_p = acceleration of pleximeter

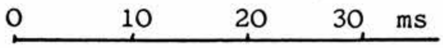
α_r = acceleration of adjacent ribs

At the ear, the sound pressures sum together to give $p = k_1\alpha_p + k_2\alpha_r$

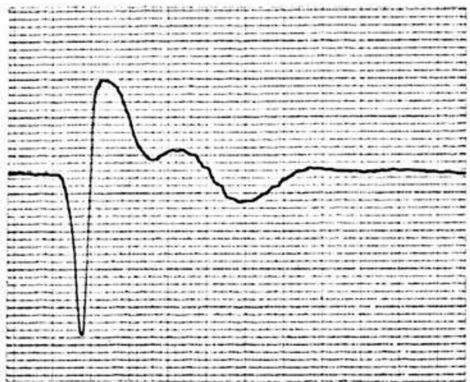
To check the validity of the model, accelerometers were attached to both the pleximeter and an adjacent finger. Then the two outputs were summed as shown in figure 4.31. With the output displayed on an oscilloscope, the attenuators were adjusted until the accelerometer outputs summed in an acceptable ratio; i.e. when the waveform of the summed output approached that of a typical 'resonant' sound. The output obtained did in fact display a close similarity in waveform to that of

Figure 4.32 COMPARISON OF SOUND PRESSURE AND OUTPUT OF SIMPLE PHYSICAL MODEL

Scale



Output of Simple Physical Model



Percussion Sound Pressure

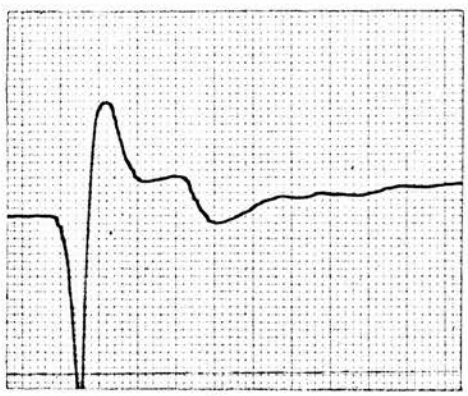
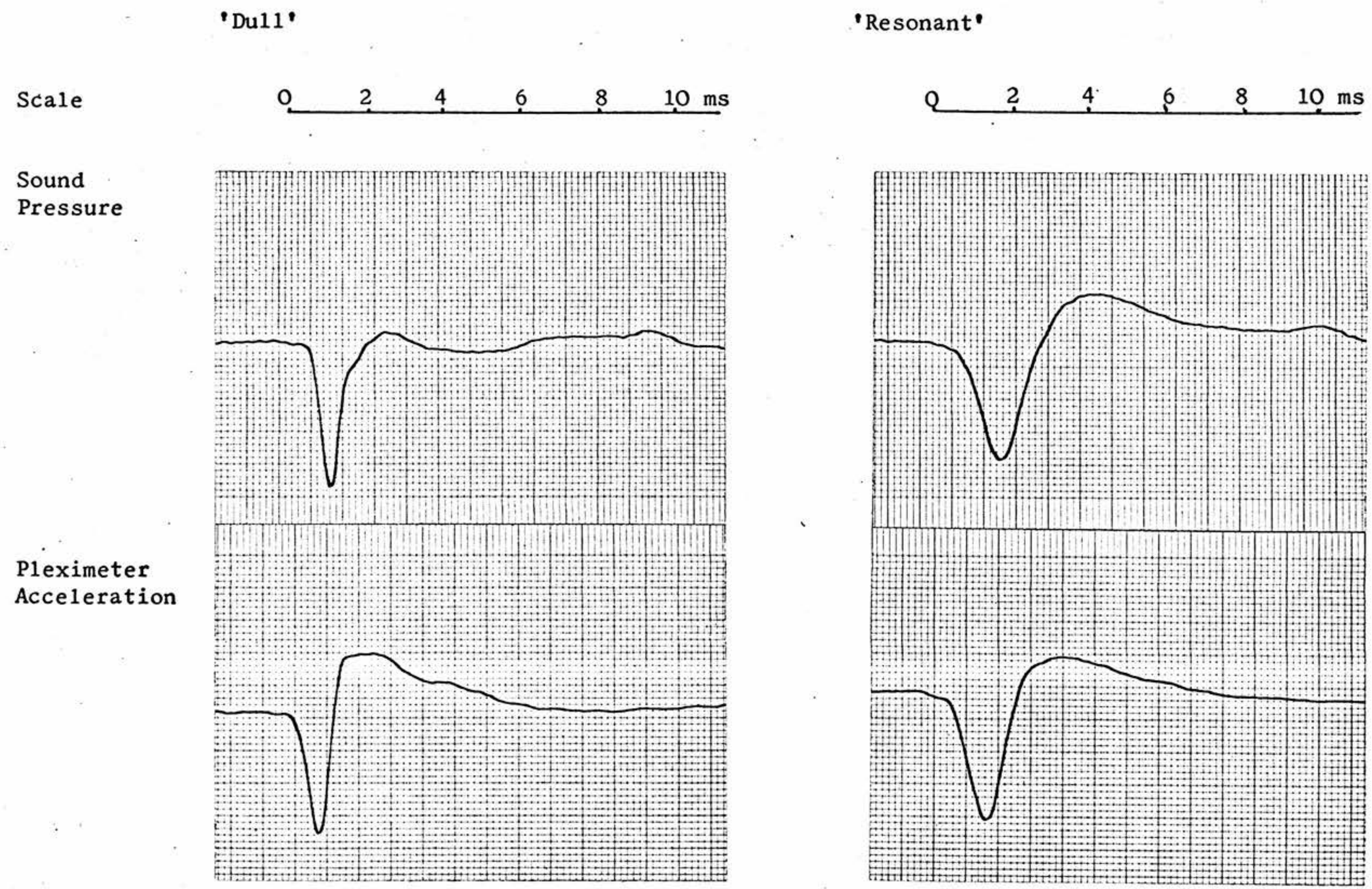


Figure 4.33 TACTILE SENSATION



the percussion sound pressure (figure 4.32).

Although a close comparison of an actual 'resonant' waveshape and the output of the model showed that they were by no means exactly similar, the simple model did work sufficiently well to provide an insight into how 'resonant' percussion sounds are formed.

Attempts to produce a 'dull' waveform from the model were unsuccessful.

4.5.5 Tactile Sensation

A comparison of the motion of the pleximeter finger over 'dull' and 'resonant' areas of the chest led to a reconsideration of the tactile sensation first discussed in section 1.6.

Figure 4.33 allows the pleximeter acceleration waveform produced over a 'resonant' area to be compared with that from a 'dull' area. The 'dull' acceleration trace was found to have a sharper waveshape than the 'resonant' acceleration trace. In particular, note the narrower acceleration spike and the sharper corner of the deceleration peak. (Acceleration was chosen to be downwards and deceleration upwards on the traces, so as to make a comparison with the sound pressure waveforms easier). The acceleration scale of the 'dull' sound has been contracted to allow for twice the full scale acceleration change of the 'resonant' sound.

A large percussion force over a 'dull' area first produces a high initial acceleration and then with the low compliance of the chest wall, the pleximeter is brought quickly to rest. On the other hand the 'resonant' sound was easier to generate and the softer blow resulted in a lower pleximeter acceleration. The high compliance of the air-backed rib cage brought the pleximeter more easily to rest; as revealed by the smoother contour of the 'resonant' acceleration waveshape.

All the acceleration waveforms were found to be simple overdamped curves with no tendency whatsoever towards oscillation. No 'vibrations' (discussed in section 1.6) could be detected in any of the acceleration curves. Hence it is suggested that the misleading use of 'sense of vibration' be abandoned when the tactile sensation is being described or discussed.

Cabot's (1906) description of the tactile sensation was found to be more apt. Therefore it is suggested that the terminology - 'sense of resistance' - of the original investigators be returned to.

'Dull' percussion areas would then simply be described as producing a high hard resistance, and 'resonant' areas, a firm springy resistance.

4.5.6 Intensity and Duration of Percussion Sounds

Out of the consideration of the pleximeter acceleration and tactile sensation arose an assessment of the qualitative description of percussion sounds presently in use. The three characteristics by which the medical practitioner most often describes percussion sounds are frequency, intensity and duration. Frequency content has already been discussed (section 4.5.4). This present section is a convenient place for discussing the latter two characteristics.

4.5.6.1 Intensity

Intensity on its own does not reveal a great deal about a percussion sound since the intensity depends, to a great extent, on how hard the physician percusses. The additional information required for making the intensity meaningful is derived from the tactile sensation.

If equal percussion forces were to be used, a 'dull' sound would appear 'soft' compared with a 'resonant' sound, which would be described as 'loud'. Hence the use of the relative terms, 'soft' and 'loud'.

No attempt is made by the physician to use a constant force, because his experience allows him to combine the sound pressure level he hears and the force of percussion he feels from the tactile sensation, to produce an assessment of relative intensity between the sounds he is comparing.

4.5.6.2 Duration

Although the main reason for using duration as a criterion of percussion sounds is more clearly seen from the sound pressure recordings, the tactile sensation does add some information.

When a 'dull' sound is generated, the pleximeter is quickly accelerated and decelerated. Both acceleration and deceleration are slightly slower with a 'resonant' sound. Hence the impression reached from the tactile sensation is that the 'dull' sound is slightly shorter.

However, the sound pressure waveform strongly suggested that duration was assessed audibly since the 'dull' sound appeared to have only one feature of any importance - a rarefaction spike. This spike was only a few milliseconds long and therefore much shorter than the 15 ms of the 'resonant' waveform. The listening tests of the following section bore this out by proving that acoustically, the 'dull' sound consisted of the rarefaction spike and very little else.

4.6 <u>DETERMINATION OF ESSENTIAL AUDIBLE FEATURES OF WAVESHAPE</u>

4.6.1 <u>Listening Test</u>

Up to this point no quantitative description of 'resonant' or 'dull' sounds had been attempted. Towards this end, it was necessary to determine which features of the waveshape should be described. Since it was not possible to give a complete description of the waveshape in any simple way, it was important that the most distinctive features be extracted for later quantitation. For this task it was decided that the experienced discriminating powers of the ear should be used to discover which features of the waveshape the ear required for the recognition of the particular quality of a percussion sound.

As already discussed, the latter part of the percussion waveshape is seriously degraded by noise and sound reflections, whereas the first part is reasonably immune due to the large pressure variations encountered. This suggested that the ear might place far more weight on the initial part of the waveform than on the 'tail'. As far as the ear was concerned, could the 'tail' be neglected? This could not be assumed as it appears that the ear performs some type of correlation analysis, which perhaps could have extracted some useful information from the 'tail'. Lange (1967), while reviewing some of the literature on this subject, discusses the possibility of both auto-correlation and cross-correlation with patterns stored in the memory .

Therefore, despite the fact that even normal reflection so masked the sounds that the latter part of the waveform could not be identified and that these sound reflections had very much smaller sound pressure variations than those due to quiet room noise, it was still necessary to determine if the 'tail' of the waveform carried any important information.

To investigate this aspect of the analysis, various portions of the percussion sound waveshape were listened to separately. As this involved artificial generation of the necessary waveshapes, the sounds produced had to be listened to through the electrostatic headphone system described in the previous chapter. The voltage waveforms driving the headphones

Figure 4.34 FUNCTION GENERATOR WAVEFORMS

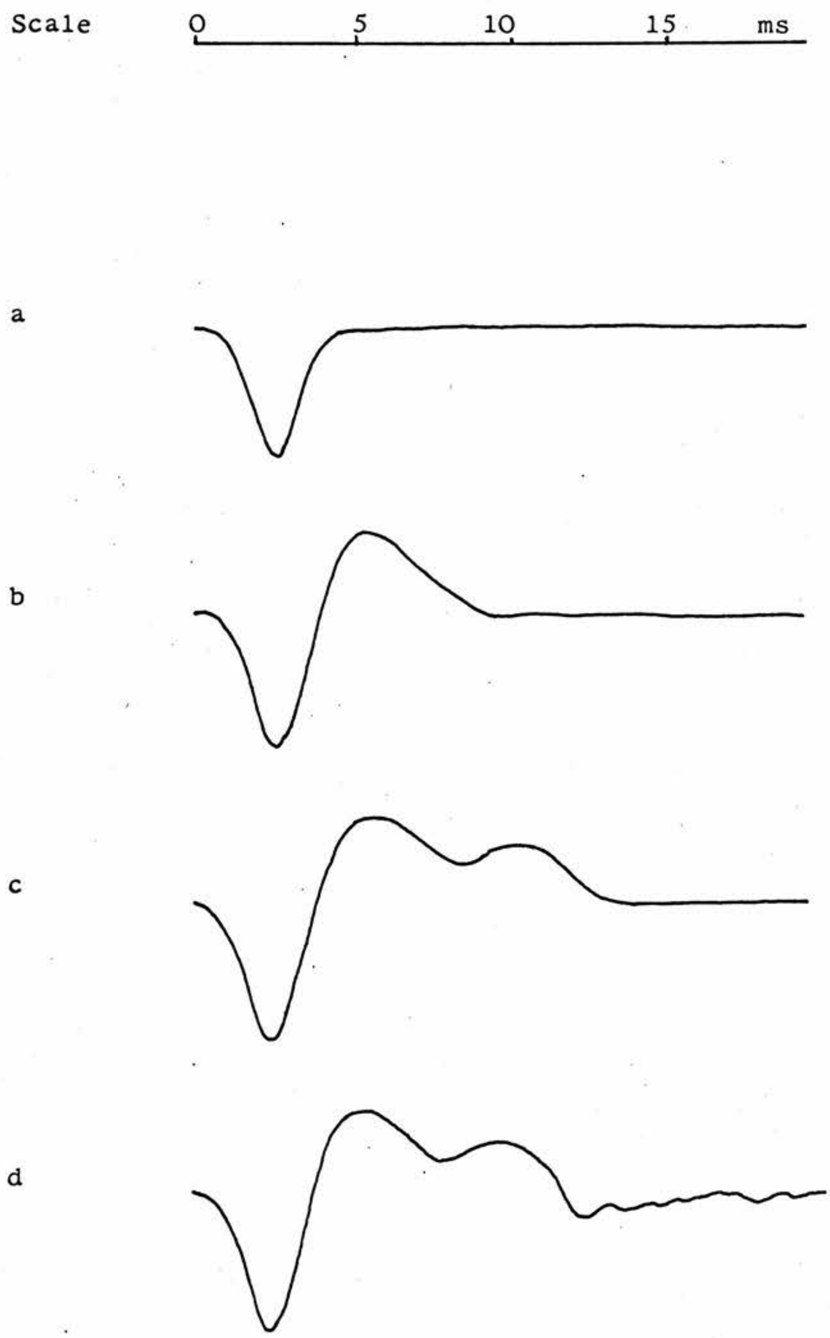


Figure 4.35 FUNCTION GENERATOR

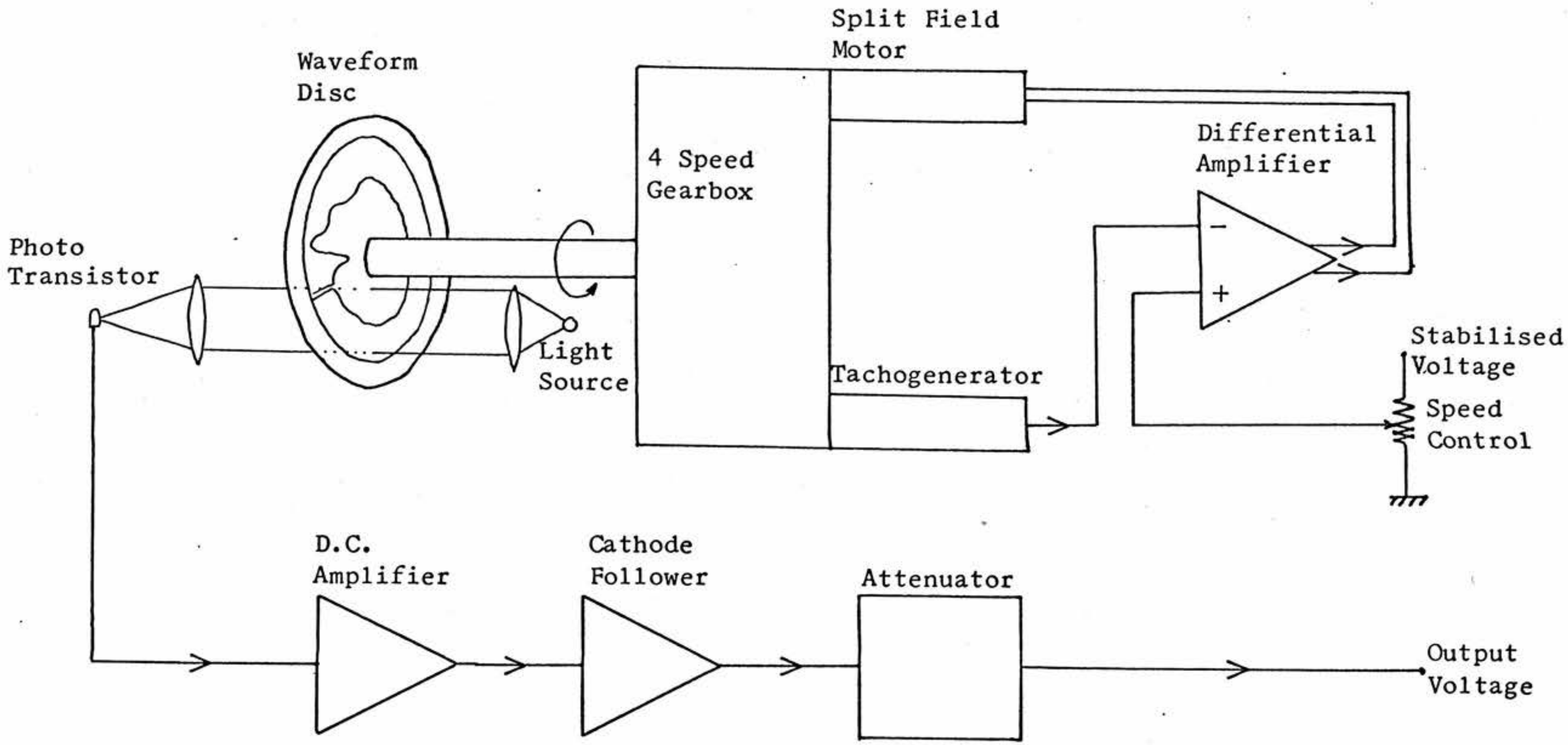
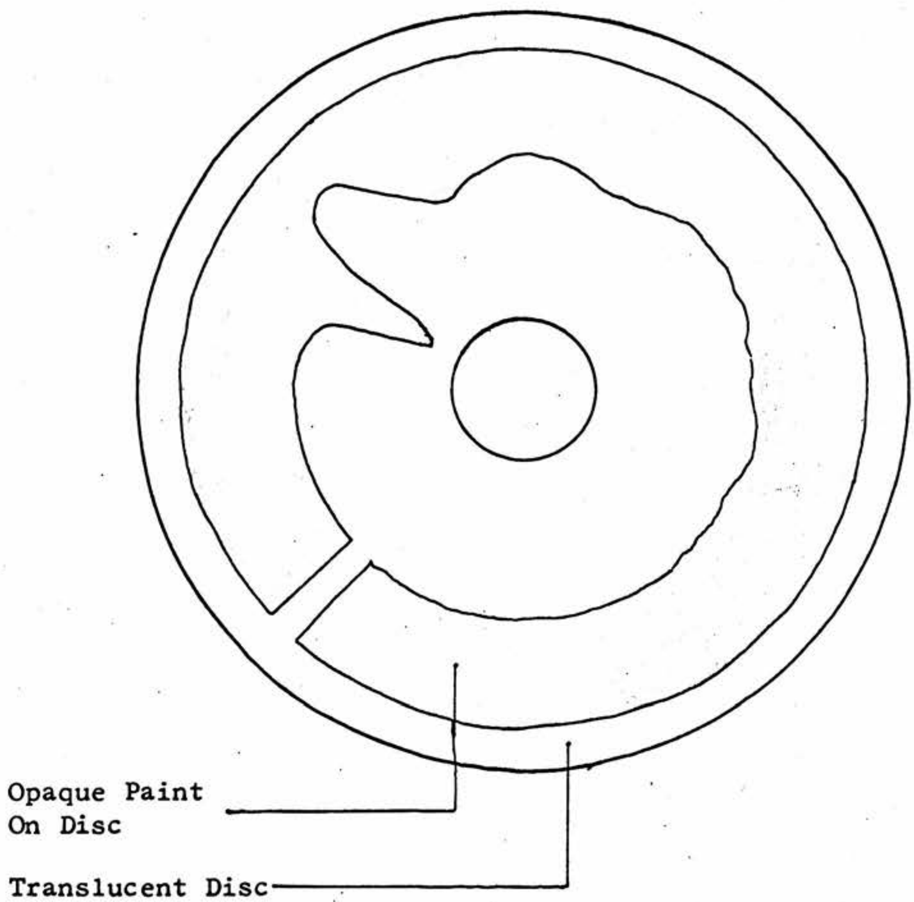


Figure 4.36 FUNCTION GENERATOR DISC



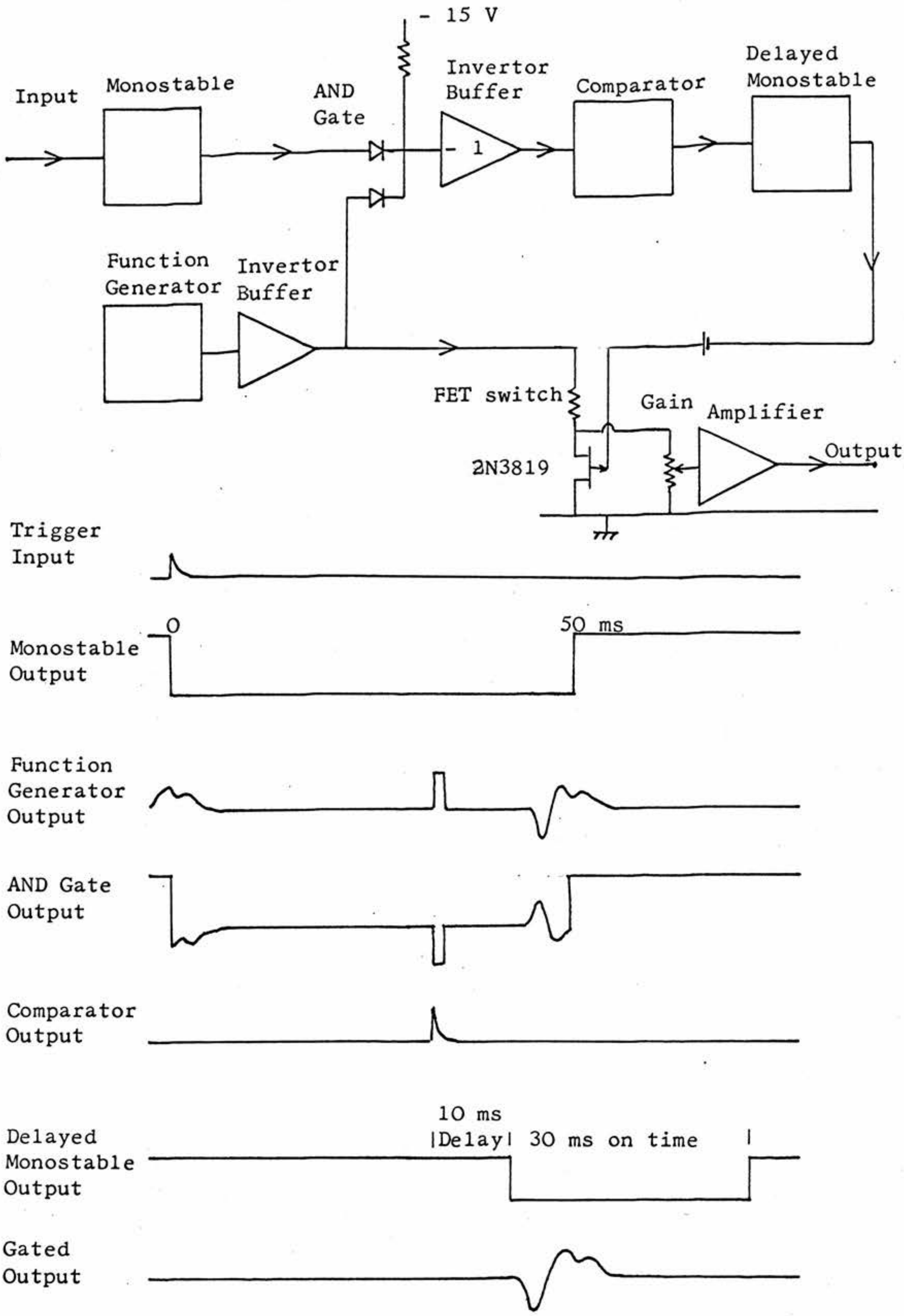
were derived from a system incorporating a special function generator (Advance SG 88). Four voltage waveforms shown in figure 4.34 were produced. Each waveform incorporated more of a typically 'resonant' percussion waveform than the previous one until that of figure 4.34d corresponded to an actual waveform complete with reflections.

4.6.2 Function Generator

Figure 4.35 is the diagram of the function generator used. This particular generator had the necessary feature of being able to reproduce, within its frequency limitations, any desired waveform. Each waveform to be produced was drawn in polar coordinates on a disc of thin Perspex. An example of a finished disc is shown in figure 4.36, where it can be seen that a section of the disc above the waveform is covered over with opaque paint. The generator worked on the principle that the output voltage was directly proportional to the light flux allowed to pass through a narrow radial slit in front of the disc to a photo-transistor detector on the other side. To obtain the voltage waveform the disc was rotated by a servo controlled motor and four speed gearbox, the speed being set by an external control.

With the disc rotating at a constant angular velocity (20 rev/s in this case) a repetitive waveform was produced at the output of the generator. When this waveform was applied to the headphones only an indistinct burr was produced which bore no resemblance to any percussion sound. The speed of rotation could have been slowed down to produce, say, one waveform per second, but this would have necessitated the impossible task of drawing the percussion waveform on less than one-sixtieth of a revolution (less than six degrees of arc). Some other solution to the problem had to be found. This was obtained by gating

Figure 4.37 GATING OF FUNCTION GENERATOR



out one cycle of the continuous waveform when it was required.

Shown in figure 4.37 is the gating circuitry developed; accompanying the diagram are the appropriate voltage waveforms to explain its operation. After an input trigger pulse was applied, the monostable flipped over into its unstable state and stayed there long enough (50 ms) to allow the first subsequent triggering pulse on the continuous waveform to pass through the AND gate and switch the comparator. Following a 10 ms delay another monostable forced the FET switch to open circuit for long enough to allow the oncoming percussion waveform to be gated out. Hence after the input had been triggered the waveform on the next complete cycle was always gated out.

4.6.3 Results

Each of the four waveforms of figure 4.34 were fed in turn to the electrostatic headphones and six subjects, non-physicians as well as physicians, were asked to compare the four outputs with an actual 'resonant' sound. The subjects had to be given a little time to become accustomed to listening to the sounds through the headphones. This applied especially to the physicians who had become accustomed to producing their own percussion sounds, and hence were deprived of the additional information contained in the tactile sensation.

The listening tests yielded the following results.

Sounds reproduced from the waveform of figure 4.34a were not felt to possess any 'resonant' quality whatsoever; in fact, they assumed a 'dull' quality. All subjects agreed on this observation. From a study of the sound waveforms this result had been anticipated - acoustically, the 'dull' sound primarily required the rarefaction spike.

Waveform (b) contained one more feature of a 'resonant' sound than

(a) - the first compression peak. This time the sound was judged by most of the subjects to have a noticeably 'resonant' quality, although still not truly 'resonant'.

As for the final two sounds (c and d), both were easily identified as being 'resonant' sounds. At the same time no difference could be distinguished between them. Again, all subjects agreed.

Hence it can be concluded that very little of the percussion information is contained in that part of the waveform after the second compression peak. The essential audible features of the 'resonant' sound contained no more than the large rarefaction spike followed by the two rounded compression peaks; the total sound duration being something in the order of 15 ms.

CHAPTER 5

P A T T E R N   R E C O G N I T I O N

5.1 THE PROBLEM

Chapter 4 dealt with the qualitative description of percussion sounds. This chapter leads on to a quantitative description. The preceding work produced a general understanding of the nature of the sounds and provided sufficient initial information to allow a decision on the subsequent development of the investigation to be made. A useful understanding, however, requires the ability to make measurements on the various sounds, which would render them suitable for comparison.

It is necessary to describe percussion sounds in this quantitative way because of the large variety of sounds which can be produced: a fact which is not immediately apparent from the use of present day terminology where the various percussion sounds such as 'hyper-resonance', 'resonance', 'impaired resonance' and 'dullness' are used in such a way as to imply that every sound falls naturally into its own specific group. However, all of these types as already noted do in fact merge with each other.

Note has previously been taken of the sounds being not well described, let alone defined, so any proposed measurements would improve the existing situation by removing the uncertainty over the description of the sounds and hence prevent more than one name being used to describe one sound; an error prevalent even in medical text books (see section 1.3). The totally subjective description would then disappear, resulting in a greater uniformity of description.

Ideally, any measurements taken should refer directly to some feature of the human body so that a measure of the patient's condition could be made. This type of analysis was begun in section 4.4. when the sound pressure waveshape obtained was correlated with the motion of the ribs near to the percussion area. The next step could have been to try and associate the rib movement with, say, the compliance of the localised volume of lung tissue; this compliance being above the normal over an emphysema or below the normal over a lesion. Unfortunately, the human body is such a complex system that this could not be followed up. Admittedly, we all have the same organs in roughly similar positions, but there is such an immense difference in shape and size of both the human body and of the internal organs that it becomes quite impossible to produce a simple model of any particular part of the thorax with an accuracy sufficient for determining results. Therefore, no further attempt was made to associate the percussion sound waveform with any measurable feature of the human body. Hence all subsequent measurements were made only on the pressure wave itself, for later correlation with known features of the human body, such as lung volume or simply the position of the percussion area on the body.

Even the taking of measurements from the waveshape posed various problems, the principal one being the variability of the waveshape itself. Many of the aspects which have some bearing on the sound pressure waveform have already been discussed in chapter 4, expecially in section 4.1. From the work contained there it can be noted that exact repeatability of waveshape cannot be expected even from two consecutive blows on the same position of the same patient, since neither will the lung volume have been kept constant nor will the physician have retained an exactly similar pleximeter firming force or plexor blow, whither in the magnitude or direction of the blow or in the time taken to withdraw the plexor after the percussion blow. Fortunately these variations only produced small changes in the percussion waveshape and since there were consistent features to be noted in each type of waveshape, these unrepeatable changes were neglected for the purpose of making a quantitative description of the sounds.

Hence the problem became that of a search for a consistent waveshape pattern in the various percussion sounds such that those sounds could be recognised in terms of this specific pattern; so enabling the characteristic features of the pattern to become measurable and a model produced.

Use had already been made of the ear in ascertaining the basic pattern of the sound waveform and this was followed up in the next section by the formulation of a model of the waveform pattern to enable the pattern to be quantitatively defined.

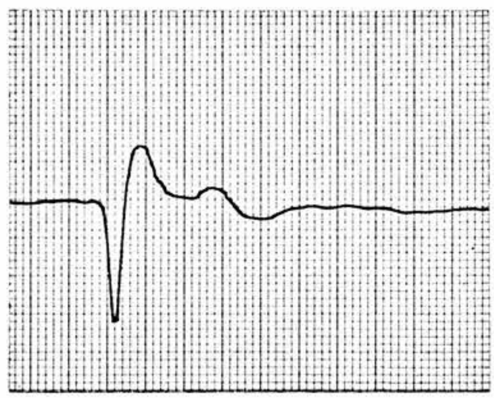
As already noted the major requirement of the 'dull' waveform was a single rarefaction spike, whereas the 'resonant' waveform required compression peaks as well. Even with those peaks the total duration was less than 15 ms. The task was then to reproduce those features on the model.

For the purpose of waveform description, 'dull' sounds with their single rarefaction formed a subset of the sounds with a characteristically 'resonant' waveform, since they also contained a single rarefaction. Following the rarefaction, the 'dull' sounds

Figure 5.1 STRAIGHT LINE APPROXIMATION TO 'RESONANT' PERCUSSION WAVEFORM

Scale

Pressure Waveform



Straight Line Approximation

contributed, no additional significant features, whereas the 'resonant' sounds did. Therefore consideration was first of all given to 'resonant' sounds, and with the principal features of the 'resonant' waveform pattern in mind a search was conducted for a function which could be used to describe the pattern.

5.2 FOUR PARAMETER MODEL OF WAVESHAPE

It is possible to build up any desired waveform from an infinite series. For example, a power series or a Fourier series. Infinite series can be truncated, since the latter terms do not contribute any significant feature to the waveform. However, even a truncated series still required a considerable number of terms in order to approximate reasonably closely to the complicated shape of a 'resonant' percussion waveform.

Since a series of n terms has n independent coefficients, a description of a waveform using such a series requires the specification of n parameters. Hence long series are not usefully applicable to the problem of percussion waveform representation.

It was necessary for the function to contain as few parameters as possible, and desirable that those parameters should measure certain obvious features of the waveshape pattern. Each parameter should be determined by one particular feature. Hence the results of any such parameter measurement could immediately form a picture of the pattern in one's mind.

No convenient continuous function could be found, so a straight line approximation as shown in figure 5.1 was selected. An advantage of the straight line approximation which was considered to be of great importance was that the function lent itself to generation by electronic circuitry. If this straight line function is now considered as being frequency limited, the sharp corners, which give the function an infinite frequency spectrum, are rounded off. Fortunately this can be easily simulated by passing an electronically generated straight line waveform through a low pass filter. Hence the function which was used to describe the percussion waveshape was chosen to be of the frequency limited straight line variety. Consideration was given first to the determination of the parameters to be used for the straight line approximation and then later to the transfer function of the filter to be employed.

The straight line approximation shown in figure 5.1 requires a considerable number of parameters to describe its shape or pattern. For example, consider the case where the function is described by its turning points, of which it has seven. Six of them require a time coordinate (keeping one point fixed). All but three of them fall on the zero sound pressure level and require no pressure level information. However the remaining three do, making a total of nine parameters required to describe the function fully. Nine parameters were considered to be too many, but as the function was already a simplification there was no reason for not putting certain additional constraint on the function.

After close examination of a considerable number and variety of percussion sound pressure waveforms the following two conclusions were reached. The second compression peak, although never particularly well defined in shape, tended both to be symmetrical about its own centre and to lie at an approximately fixed distance from the rarefaction impulse. The second observation concerned the first

Figure 5.2 STRAIGHT LINE APPROXIMATION USING GRADIENTS

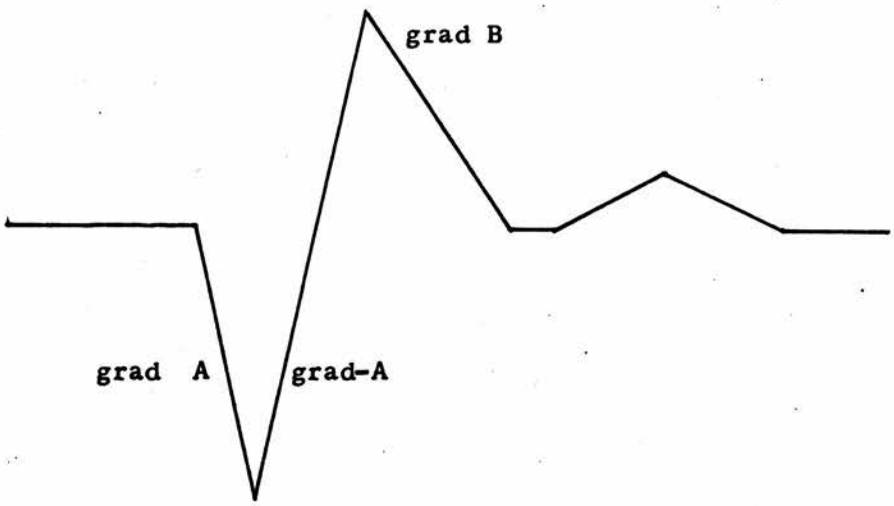


Figure 5.3 STRAIGHT LINE APPROXIMATION USING TIMES

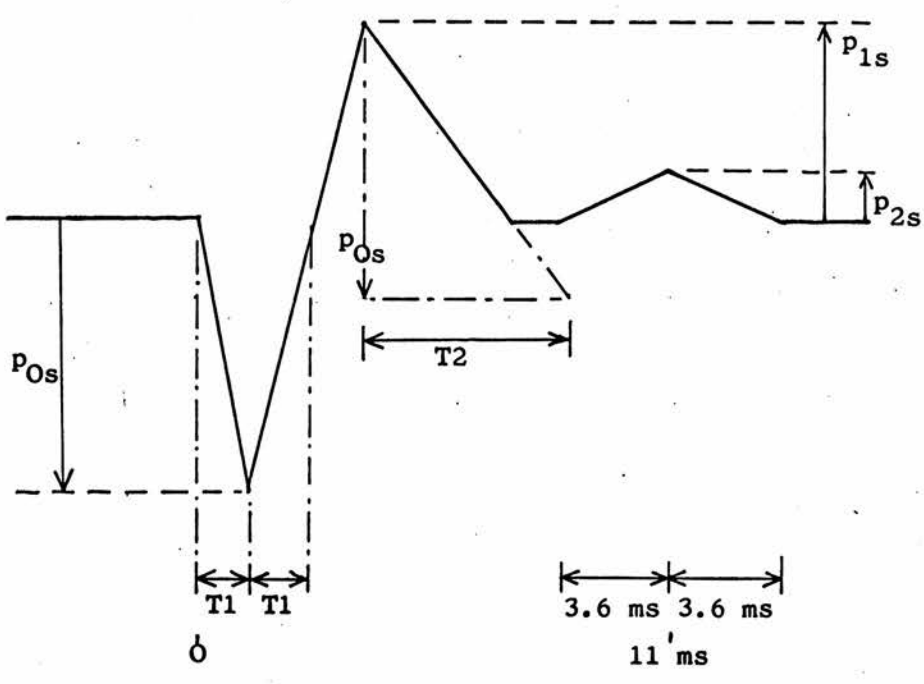
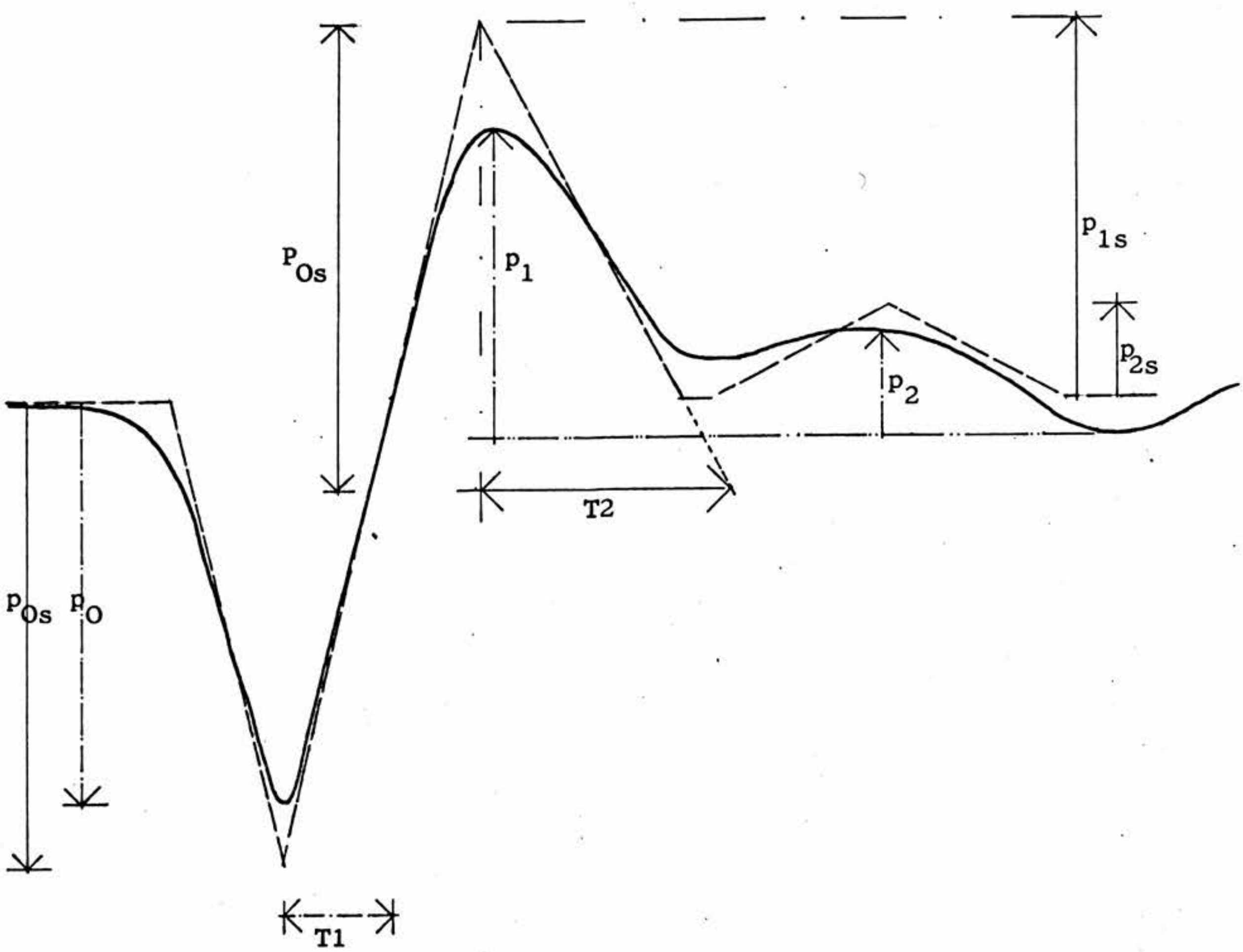


Figure 5.4 MEASUREMENT OF THE FOUR PARAMETERS FROM THE PERCUSSION WAVESHAPE



———————	Actual Waveshape
— — — —	Straight Line Approximation
T1, T2, P1 $(\frac{p_1}{p_0})$, P2 $(\frac{p_2}{p_0})$	Four Parameters

gradient of the rarefaction; the value of this gradient, although changing quite noticeably was found to be approximated to reasonably well by the negative of the second gradient. Both those approximations are shown implemented on the straight line function in figure 5.2.

One further alteration had to be made to the function. Gradients, although useful in the illustration of how the function was derived do not result in practical units since time is measured along the x-coordinate and sound pressure along the y-coordinate, and any change in recording distance from the microphone would result in changed gradients even although the waveform had only changed in size and not in waveform pattern. It was considered far better to measure one consistent unit and time was selected. The measurement of T1 is obvious from figure 5.3 since it is an actual time, while that of T2 is a projected time such that

$$\frac{T1}{T2} = \frac{\text{grad B}}{\text{grad A}} \qquad 5 - 1$$

and so the relationship between the two gradients was preserved by T1 and T2 (even if in the inverse). Therefore the ease with which the waveshape pattern could be reconstructed in the mind was retained.

The measurement of p_0, p_1 and p_2 were taken with reference to the frequency limited model function or from an actual waveform (a subscript 's' is used with the straight line approximation), but in order to prevent absolute measurements the values of the two compression peaks were rationalised and the ratios p_1/p_0 and p_2/p_0 actually measured. These two ratios will now be referred to as P1 and P2 respectively.

Figure 5.4 illustrates the measurement of the four parameters from an actual waveshape. The relationship between the heights of the peaks of the straight line function and those of the actual waveform (or the filtered model) is dealt with in the following section.

5.3 SIMULATION OF MODEL

5.3.1 Necessity of Simulation

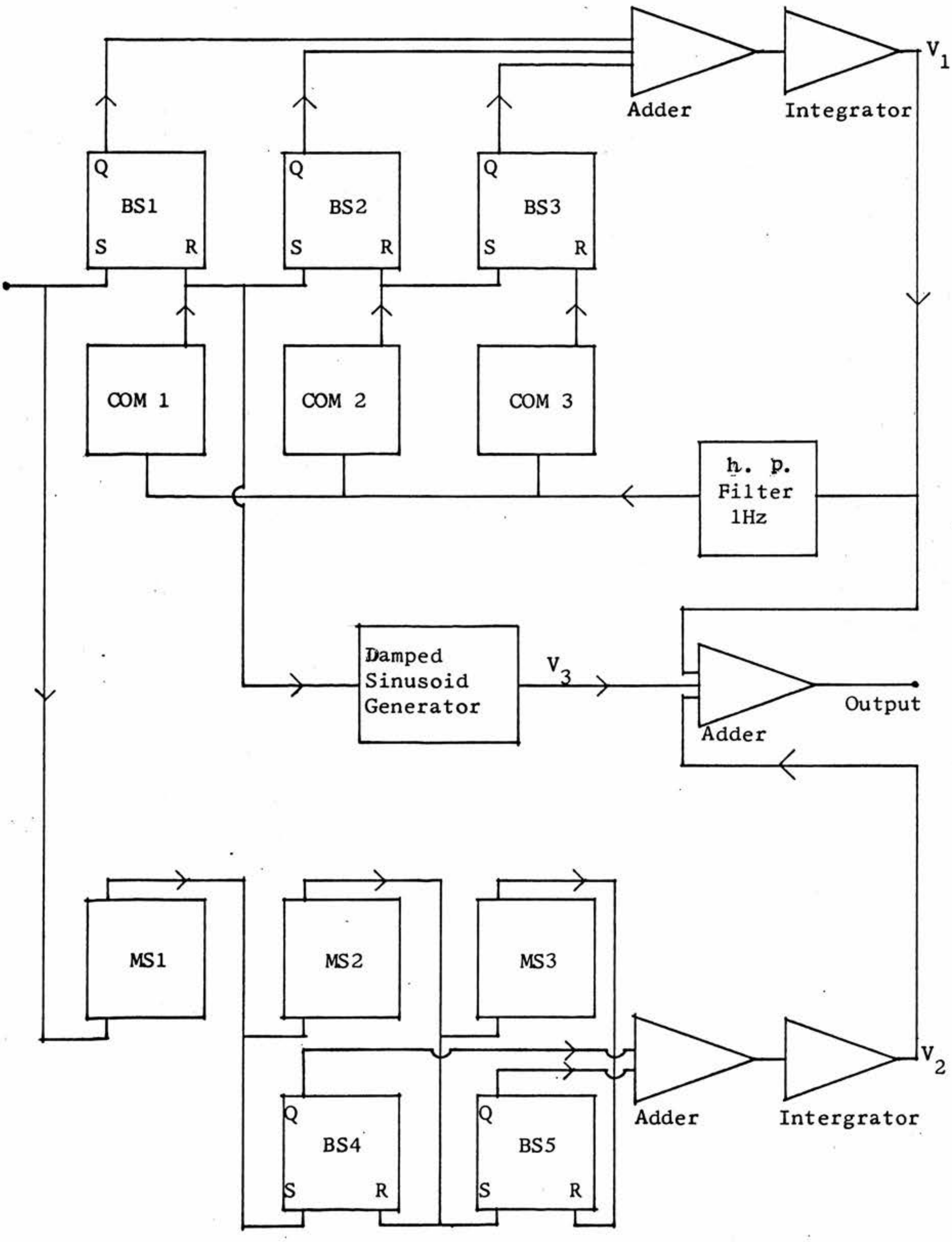
Generally, when measurements are made some information is deliberately discarded and only those measurements of particular relevance to the task on hand are recorded. This arises because some features of the system or waveshape from which the measurements are being taken are often more important than others. However, concerning this present project it was impossible to say at this stage if any features of the percussion sounds were more relevant than others or more helpful in distinguishing the various percussion sounds. Therefore it was decided not to attempt to throw away any information contained in the four parameter model, but rather to check that no significant information had already been discarded in the simplification required in producing this model.

Hence it was proposed to determine how well the original waveform could be reconstructed from the model by comparing the model with the original percussion sound. Therefore a simulator had to be built to produce the voltage waveforms of the model from its four parameters so that by connecting it to the electrostatic headphones a comparison could be made of the sounds produced as well as of the waveforms.

5.3.2 The Simulator

That each control of the simulator should set only one of the four parameters and that the value of that parameter should be completely

Figure 5.5 SIMULATOR - BLOCK DIAGRAM



BS - Bistable; COM - Comparator; MS - Monostable

Figure 5.6a SIMULATOR - SCHEMATIC DIAGRAM

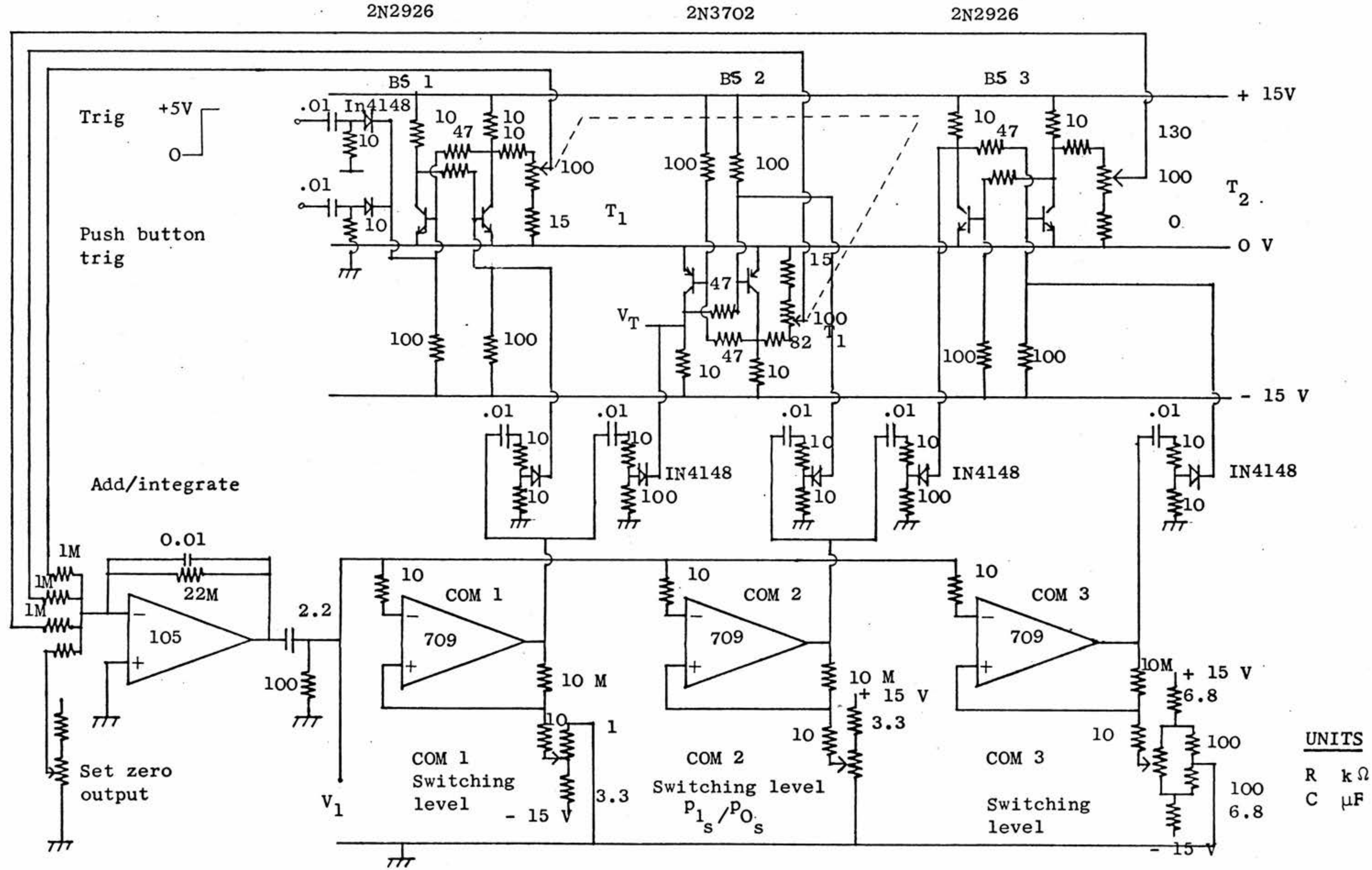


Figure 5.6b SIMULATOR - SCHEMATIC DIAGRAM

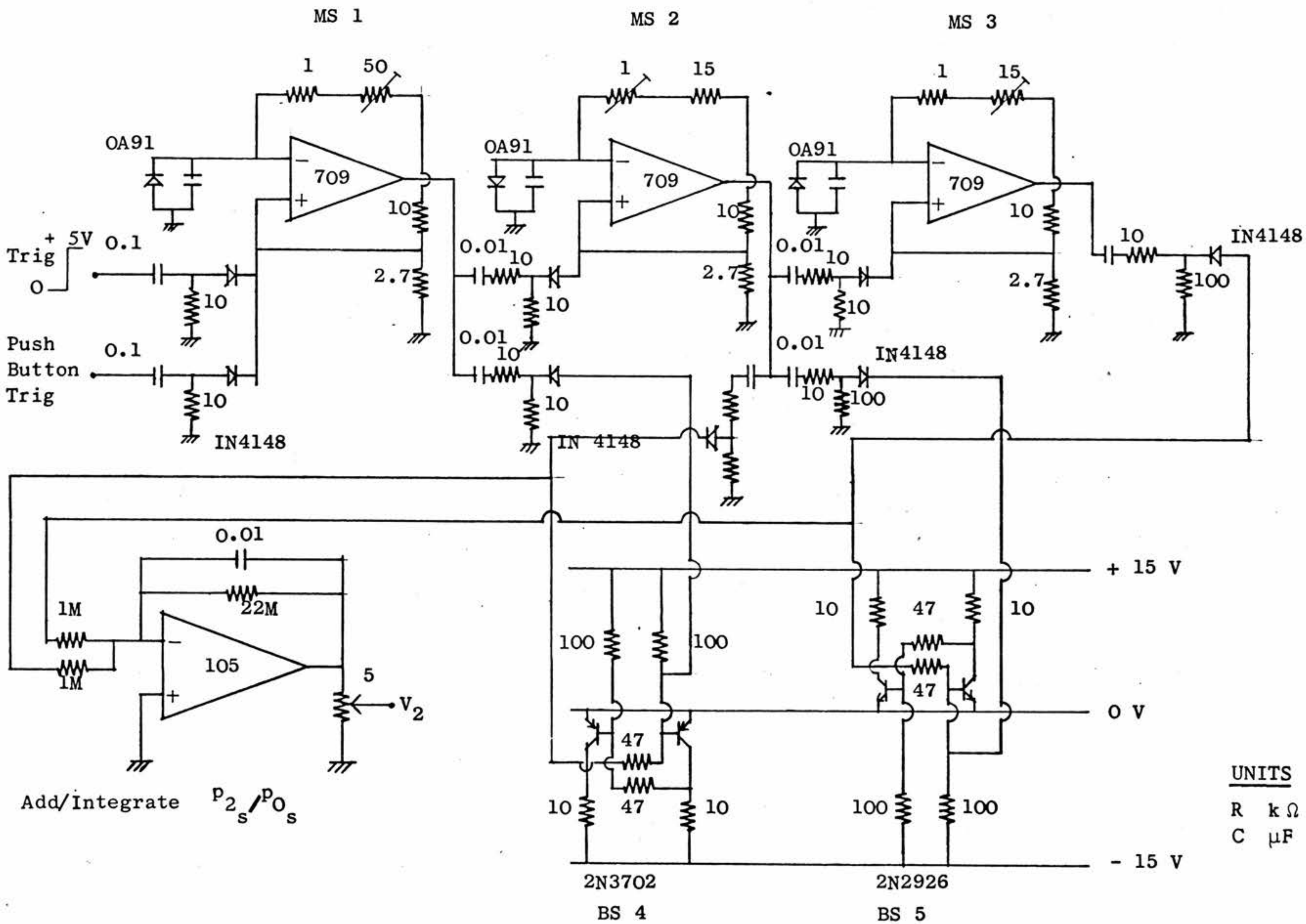
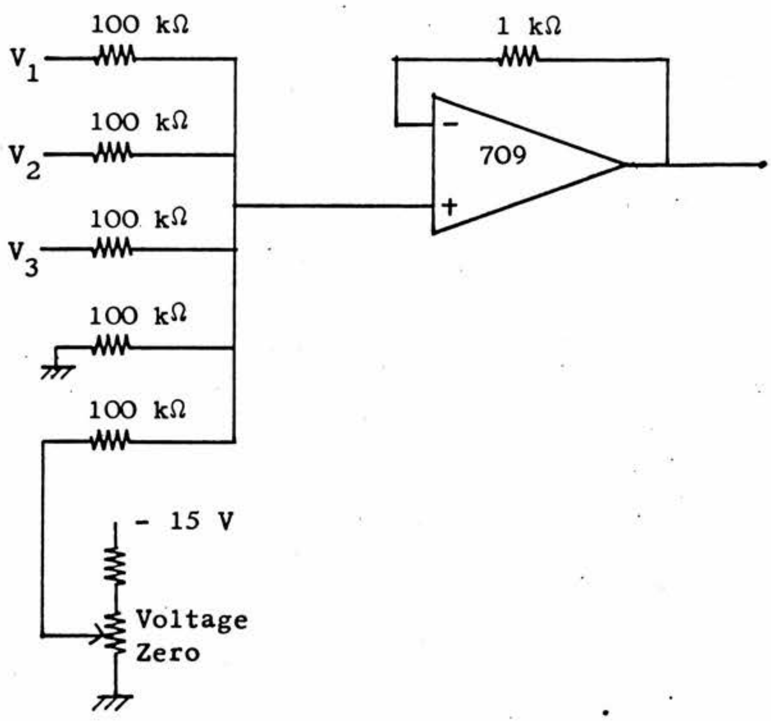


Figure 5.6c SIMULATOR - ADDER AND TRIGGER

Adder



Push Button Trigger

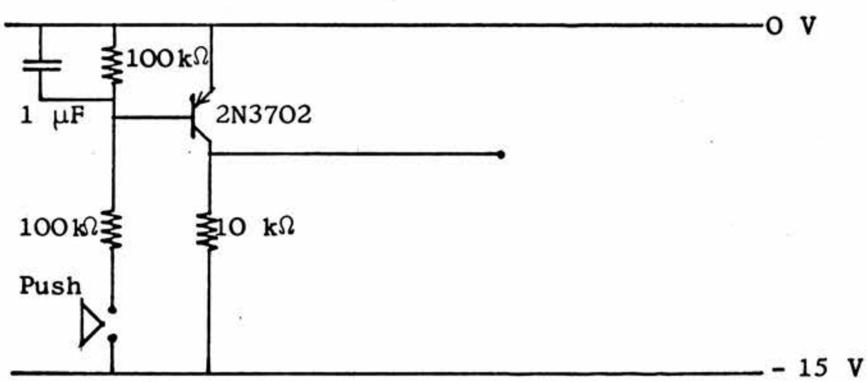


Figure 5.6d SIMULATOR - DAMPED SINUSOID OSCILLATOR

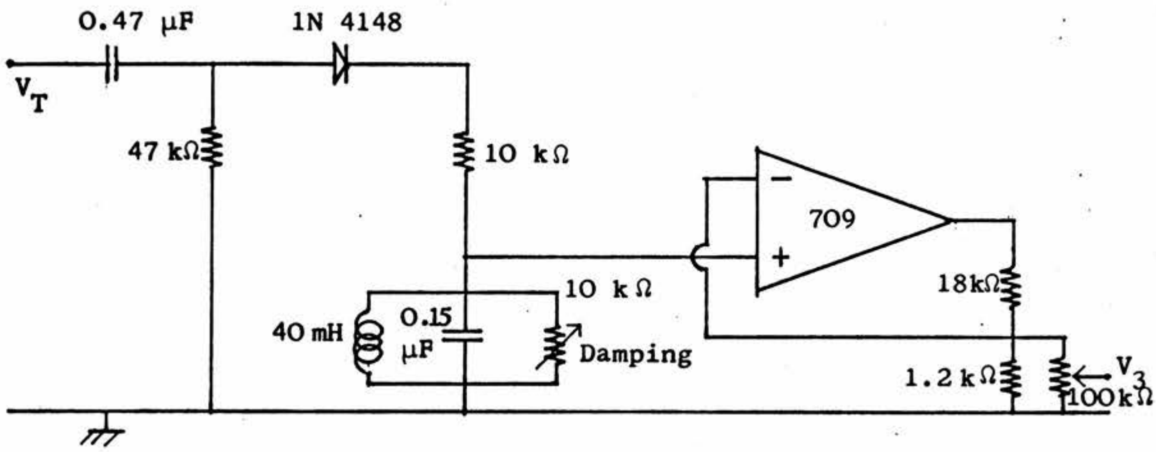
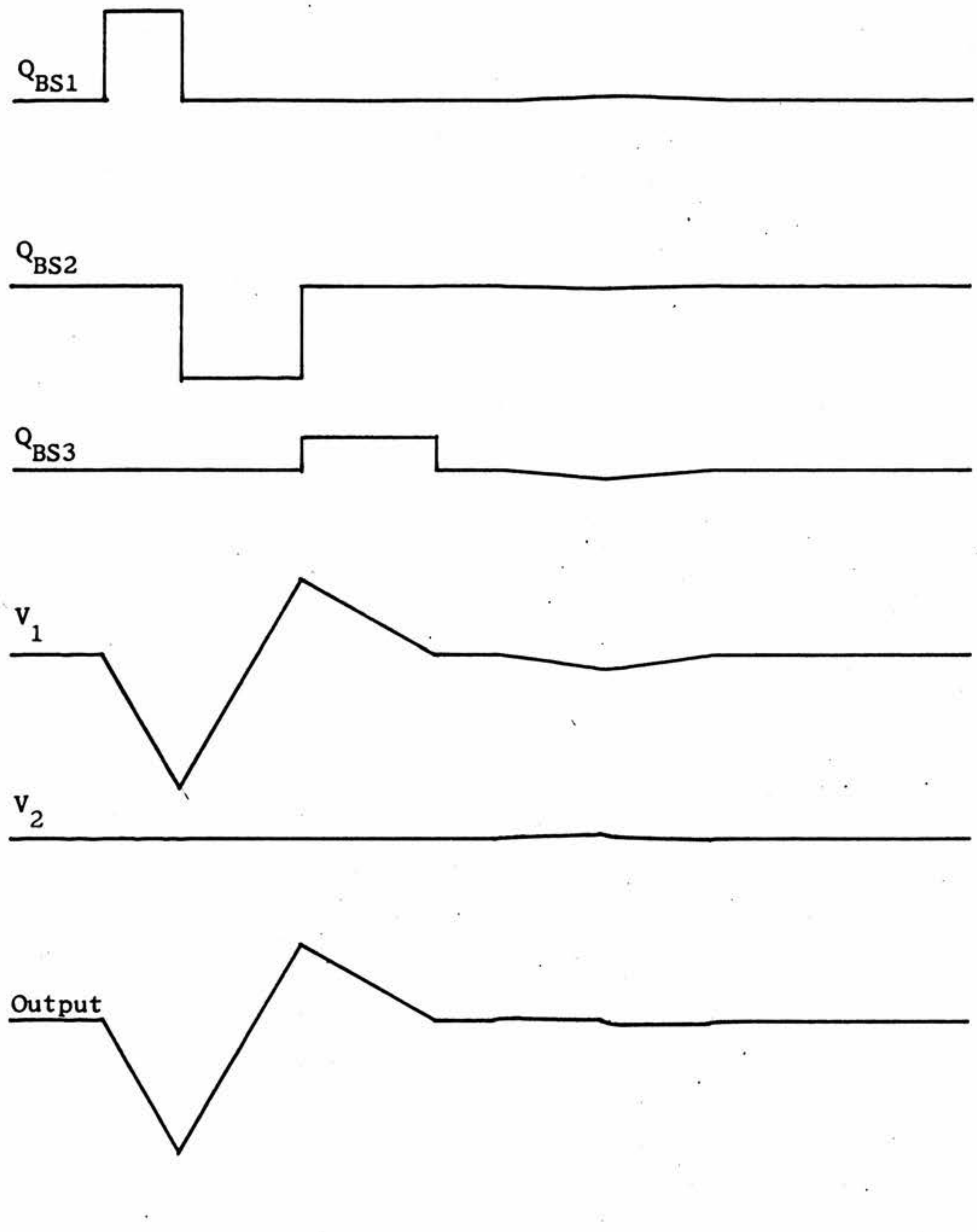


Figure 5.7 SIMULATOR WAVEFORMS



independent of the other controls, was considered to be a necessary requirement of the simulator so that each waveform could be easily set up and so that the values of the parameters of the straight line function could be read directly from the calibrated scales of the simulator.

The block diagram of the simulator which was designed and constructed is shown in figure 5.5, with its various schematic drawings in figures 5.6a to 5.6d. Both the rarefaction and first pressure peak were generated together by the circuitry of figure 5.6a, while the second compression peak was produced completely separately (figure 5.6b), and then both waveforms were combined in the adder (figure 5.6c). Also connected to the adder is the damped sinusoid generator which was introduced to simulate the resonance of the percussion fingers during the listening tests (see section 5.4.2.1). Each simulated waveform was triggered either by a remote push button switch connected to the front panel of the simulator or by an input voltage step from a low frequency (below 1Hz) pulse generator if regular triggering was required.

Various voltage waveforms from the simulator are illustrated in figure 5.7 to aid an explanation of its operation. The first three waveforms show the output of the three bistables of figure 5.6a, the magnitude of these outputs being dependent upon the attenuator settings used. These three voltages were added together and then integrated, producing the voltage V1 and so as to enable the first two gradients of V1 to be equal in magnitude, the output voltages of both bistables had also to be equal in magnitude. This was achieved by ganging the two potentiometers controlling the output of these two bistables; and the first parameter T1 obtained from the setting of this ganged potentiometer. Likewise, the second parameter T2 was obtained by controlling the attenuator potentiometer of the third bistable.

These three bistables were switched by the output of the comparators which continually monitored the integrated output (V1). The first was set to switch on an output voltage of - 0.7 V, resulting in an output voltage waveform which always turned at that voltage, so fixing the size of the negative rarefaction spike and giving a reference voltage level to which the two positive peaks of the waveform were scaled. The value of the first peak was controlled by the switching voltage of the second comparator which was set by another potentiometer whose scale was calibrated with the value of $\frac{p_{1s}}{p_{0s}}$ ($P1_s$) Resetting the third bistable after the base line had been reached, was achieved by the third comparator.

Controlling the bistables generating the second compression peak were three monostables (figure 5.6b). These monostables were preset for a fixed on-time since the position of the second peak had already been fixed. Again a ganged potentiometer controlled the output of the two bistables and its scale was calibrated to read $\frac{p_{2s}}{p_{1s}}$ ($P2_s$)

To determine the required range of the four parameters a large number of wavetraces were studied and the simulator calibrated to cover the range shown below.

Parameter	Simulator Range
T1	5 - 30 x 10^{-4} s
T2	1 - 30 x 10^{-3} s
$P1_s$	0 - 1
$P2_s$	0 - 1

As has already been noted, the sharp corners of the straight line function introduce a theoretically infinite frequency spectrum. In actual practice, when generating such a function this is never the case. However, the sharp corners do introduce a very high frequency spectrum and so by removing the highest frequency components of the straight line simulation a more realistic reconstruction of the original was obtained.

To remove as much of the high frequency spectrum as possible and yet at the same time introduce no instability a low pass filter with a sixth order Butterworth response was used. An n^{th} order Butterworth response can be described by the following formula,

$$\left| G(\omega) \right| = \frac{1}{\sqrt{1 + \left(\frac{\omega}{\omega_o}\right)^{2n}}}$$

ω = angular frequency

ω_o = - 3dB angular cut-off frequency

A sixth order filter of this type does however introduce an overshoot on fast steps, so on the filter used (Barr and Stroud, type EF2) one of the second order amplifiers was damped preventing any such distortion.

In choosing the filter cut-off frequency a compromise had to be made. With the cut-off frequency set too low the rarefaction impulse was distorted since this impulse contained the fastest rise times of the waveshapes and hence the greatest high frequency components. With the cut-off set too high, the first compression peak was not smoothed sufficiently. The compromise had to be made such that the filter produced the most realistic waveforms from the majority of

Figure 5.8 FILTER FREQUENCY RESPONSE

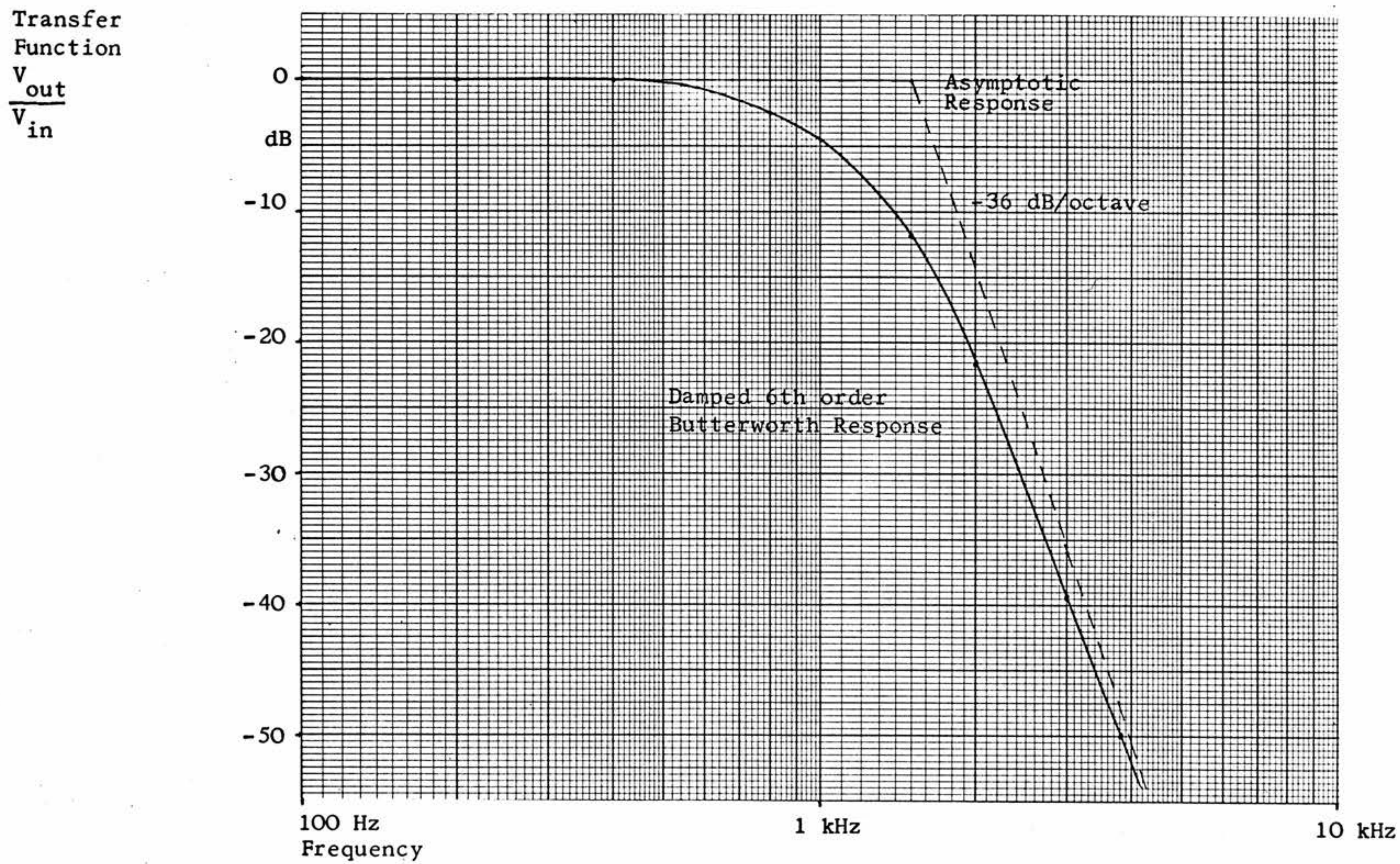


Figure 5.9 PERCUSSION WAVEFORM SIMULATOR AND ELECTROSTATIC HEADPHONE SYSTEM

straight line simulations. In view of the range of the simulations no optimum frequency could be specified. Nevertheless after considerable visual examination of the waveshapes and audible assessment of the sounds of numerous percussion simulations filtered at various frequencies, 1.5 kHz was selected as the filter cut-off frequency, although this was by no means critical.

Hence the four parameter model of the percussion sounds was finally defined as a frequency limited version of the straight line approximation where the filter had a sixth order damped Butterworth response as shown in figure 5.8.

Making use of the simulator was simple and straightforward. After measuring the values of the four parameters, T1, T2, P1 and P2 from a recording of a wavetrace, the values of T1 and T2 were first set up directly on the calibrated scales of the simulator. Values of $P1_s$ $(\frac{p_{1s}}{p_{0s}})$ and $P2_s$ $(\frac{p_{2s}}{p_{0s}})$ on the simulator do not yield similar P1 $(\frac{p_1}{p_0})$ and P2 $(\frac{p_2}{p_0})$ values measured at the filter output, so the actual values of P1 and P2 were set up by observing the filtered output on an oscilloscope and noting the simulator p ratios required to give the actual P1 and P2 values at the filter output.

Figure 5.9 shows a photograph of the simulator connected to the electrostatic headphone system.

5.4 VALIDITY OF MODEL

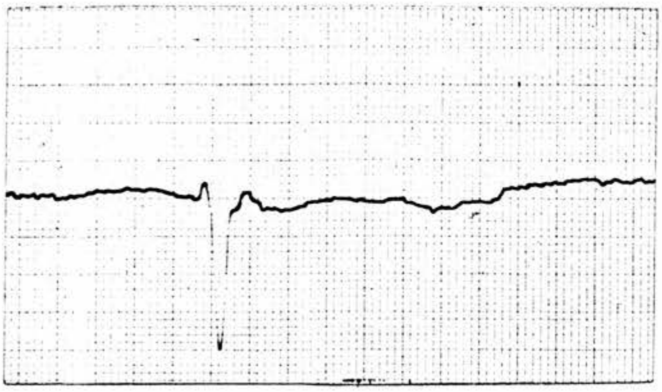
5.4.1 Visual Proof

To enable a visual comparison of the actual and simulated waveshapes to be made, four different but typical percussion sounds were recorded on the chart recorder. From these recordings, the values of the four parameters were measured, then the simulator was set up to enable the

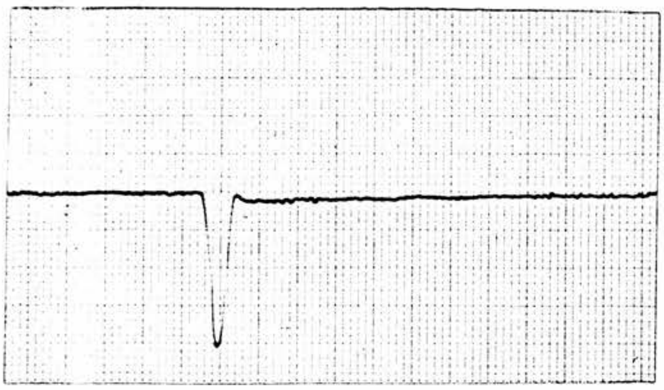
Figure 5.10 ACTUAL AND SIMULATED 'DULL' SOUNDS

Scale 0 5 10 15 20 ms

Actual 'Dull'



Simulated 'Dull'



Parameters

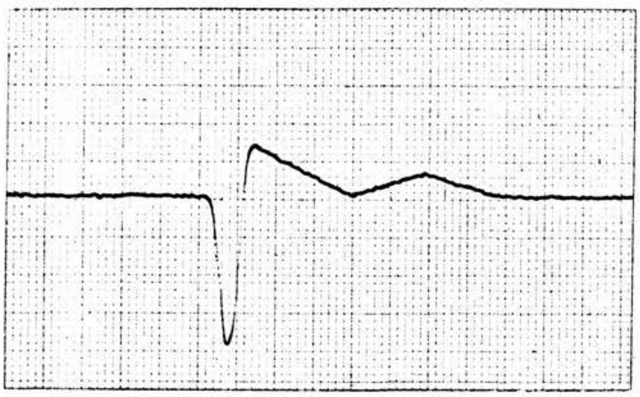
T1 (x 10^{-4})s	T2 (x 10^{-3})s	P1	P2
5	/	0	0

Figure 5.11 ACTUAL AND SIMULATED 'IMPAIRED RESONANCE'

Scale 0 5 10 15 20 ms

Actual 'Impaired Resonance'

Simulated 'Impaired Resonance'



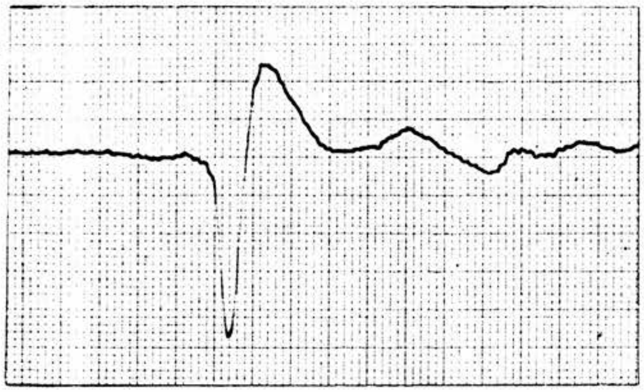
Parameters

T1 $(\times 10^{-4})$s	T2 $(\times 10^{-3})$s	P1	P2
6	12	0.3	0.15

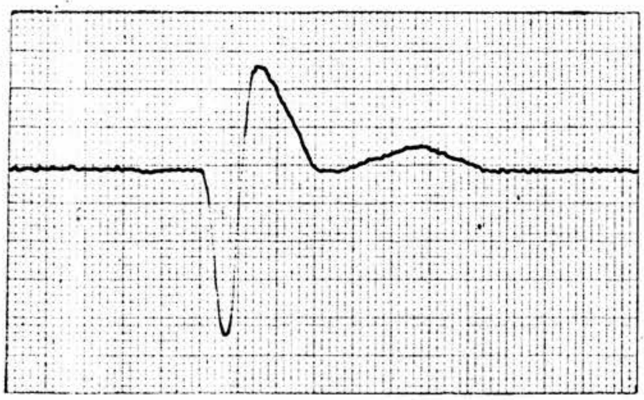
Figure 5.12 ACTUAL AND SIMULATED 'RESONANCE'

Scale

Actual 'Resonance'



Simulated 'Resonance'



Parameters

T1 ($\times 10^{-4}$)s	T2 ($\times 10^{-3}$)s	P1	P2
7.5	5	0.55	0.2

Figure 5.13 ACTUAL AND SIMULATED HIGHLY 'RESONANT' SOUNDS

Scale 0 5 10 15 20 ms

Actual Highly 'Resonant'

Simulated Highly 'Resonant'

Parameters

T1 ($\times 10^{-4}$)s	T2 ($\times 10^{-3}$)s	P1	P2
10	9	0.6	0.3

various models of the percussion sounds to be generated and these waveforms were then also recorded on the chart recorder. Each original waveshape is shown alongside its simulated version in figures 5.10 to 5.13 to enable a comparison to be made; a table of parameters is also included with each.

The 'dull' sound of figure 5.10 had only one measurable parameter but, as expected, fitted neatly into the pattern of measurement for 'resonant' sounds. This one parameter described only the rarefaction impulse and neglected all else, yet still retained the basic shape of the 'dull' sound.

With the three 'resonant' sounds all four parameters were required. The rarefaction impulse of each of these sounds was well preserved, but the first compression peak revealed the major flaw of the model; the peak of the model having a sharper turning point than the original sound. This was particularly noticeable on the smooth waveform of the highly 'resonant' sound shown in figure 5.13. Nevertheless the basic shape of each waveform had been retained and the simulations were easily recognisable to be from their own particular original waveform. So the visual analysis confirmed that the basic waveform pattern had been retained by the four parameters selected for the model.

5.4.2 Audible Proof

5.4.2.1 Finger Resonance

On listening to the various percussion sounds it became clear that there was some small feature lacking from the simulations. This was the resonance of the percussion fingers which had already been noted by McKusick (1955) to play a part in giving the 'dull' sound in particular its characteristic quality. When the plexor strikes the pleximeter, the blow sets the fingers into vibration. Percussion sounds such as 'resonance' which are generated easily do not give rise to much finger resonance because of the soft blow required and because the high compliance of the air filled chest cushions the blow. On the other hand, a far greater force is always used to produce a 'dull' sound and since no air cavity underlies the percussion area the blow meets a greater resistance and so a substantial amount of finger resonance is produced.

Adding an exponentially damped sinusoid to the waveforms to simulate this finger vibration greatly helped in adding a depth of realism to the sounds even although the sinusoid was so small as not to make any significant difference to the simulated waveshape. The presence of the damped sinusoid generator has already been noted in figure 5.6d. Listening tests were conducted to ascertain the best frequency and damping required for simulating the finger resonance and a frequency of 1.5 kHz with a 3 ms damping time constant was selected, although this was by no means critical. The damped sinusoid can be expressed in volts by

$$K_s \cdot \sin (2\pi f_s t)\, e^{-t/\tau_s} \cdot u(t)$$

where f_s = frequency of simulation = 1.5 kHz

τ_s = time constant = 3 ms

$u(t)$ = unit step

K_s = constant

This must not be taken to suggest that the exponentially damped sinusoid is the actual waveform generated by the finger resonance, but only that it can be approximated to quite simply by this waveform. The actual waveform must be a combination of a number of different waveforms coming from various sections of the fingers. Nevertheless the simple expression was perfectly adequate since it was used only to add that extra realism to the waveforms during the listening tests.

When a value for simulated finger resonance is expressed in the listening tests it is the value of K_s in the above expression; that is before the waveform is filtered along with the straight line function. At 1.5 kHz the output response of the filter is 12 dB down and so reduces the size of the sinusoid actually listened to.

5.4.2.2 Listening Tests.

For an audible comparison of real and simulated sounds, three different types of percussion sounds were first recorded on magnetic tape. The sounds comprised a 'resonant' sound from the fourth intercostal space, 'impaired resonance' from the fifth intercoastal space and a 'dull' sound from the sixth intercoastal space with no lung underlying. From each of these three types of real sounds, four typical parameters were measured to enable them to be simulated. Then four real sounds and four simulated sounds of each type (making twenty-four in all) were recorded in a random sequence on the tape.

Listed below in the table are the values of simulation parameters used in each case.

Percussion Sound	T1 x 10^{-4}s	T2 x 10^{-3}s	P1	P2	Simulated Finger Resonance (K_s)
'Resonance'	11	9	0.5	0.2	0.1
'Impaired resonance'	7	5	0.3	0.1	0.3
'Dull'	5	-	0	0	0.5

Table 5.1 RESULTS - PERCUSSION SOUND IDENTIFICATION

Subject	Sound	a	b	c	d	e	f	g	h	i	j	k	l	m	n	o	p	q	r	s	t
1	Rr		I	I	I							I		I	I		I	I	I		I
2	Rs												I				I				I
3	Ds													I		I			I		
4	Ir								R						D	R			D		
5	Dr				I														I		
6	Is				R										R		D				
7	Dr																				I
8	Ds			I					I					I	I						I
9	Rs																				
10	Rr			I										I							
11	Is					D		D											D		
12	Ir		D	D			D			D							D		D		R
13	Rr											I									
14	Is	D	D	D		D					D		R				D				
15	Rs			I												I	I				I
16	Dr			I	I																
17	Rr			I	I									I							
18	Dr	I							I		I		I	I	I						
19	Rs			I	I					I							I				I
20	Ir						D			D						D				D	R
21	Ds												I								
22	Is	R			D	D							R								
23	Ir				D	D								D			R		D		
24	Ds					I			I										I		

R = 'Resonance'
I = 'Impaired Resonance'
D = 'Dullness'

r = real
s = simulated

N.B. Only the wrongly identified sounds are noted

In the test, the listeners, both physicians and non-physicians, were asked to identify the types of sound heard without previously having been informed that some were simulated. The tape recorder was stopped after each sequence of sounds had been heard. Each sequence corresponded to one sound repeated six times so as to allow the listener sufficient time to become accustomed to the sound. The subject was then given a chance to compare the sounds heard with actual sounds, which were percussed from the chest, before being asked to make a final decision on the sounds heard. If requested the tape recording was repeated.

Subjects who identified more than five real sounds wrongly were eliminated from the following analysis as it was felt that their judgment could not be relied upon. The results of the first twenty suitable tests are shown in table 5.1, from which the random order of the sounds can also be seen. On the table the latter 'R' denotes 'resonance', 'I' 'impaired resonance' and 'D' 'dullness'; the subscript identifies each as being either real (r) or simulated (s). To prevent cluttering up the table only those sounds identified wrongly were entered on the table.

The object of the analysis was to see if the subjects could identify the simulated sounds as well as they could identify the real sounds and hence show that the simulations contained adequate information to describe each sound such that the various sounds could still be differentiated. If people had not been able to identify the simulations or if their error rate had been higher than with the real sounds, the simulations could not have been considered as adequate representations of the real sounds. However, on scanning the results

Table 5.2 IDENTIFICATION OF PERCUSSION SOUNDS

Real Sounds

Actual Real Sounds	Identified as: D	I	R
D	69	11	0
I	16	59	5
R	0	16	64

Simulated Sounds

Actual Simulated Sounds	Identified as: D	I	R
D	68	12	0
I	12	63	5
R	0	12	68

D = 'Dullness'
I = 'Impaired Resonance'
R = 'Resonance'

this did not seem to be the case, and the following section analyses the results rigorously to show that in actual fact there was no significant difference between the errors made in identifying real and simulated sounds.

This experiment did have two drawbacks, one was due to the tape recorder noise and the other arose because of the unnatural way of listening to percussion sounds through headphones; for the medical practitioner the latter was the most objectionable feature. The tape recorder noise was apparent because the dynamic range of the recorder was limited to 40 dB, which although good for frequency modulated tape recorders, was considerably less than the 120 dB dynamic range of the ear.

5.4.2.3 Analysis of Results

Table 5.2 shows the combined results of all twenty subjects, with the real and simulated sounds separated out into different tables. The numbers in the tables show the total number of each type of sound identified in the three possible ways ('resonance', 'impaired resonance' and 'dullness'). If each sound had been identified correctly the figure 80 would have appeared in the three boxes of the downward sloping diagonal and all other boxes would have shown zero. In fact these diagnonals do hold the highest numbers and also zeros do appear in the opposite corners showing that no real or simulated 'resonant' sound had been identified as 'dull' or vice versa. Hence it can be seen that the number of correctly identified sounds (on the diagonals) was high and that the same type of error had been made in both cases.

This again suggested that there was very little difference between the errors with real and simulated sounds and a quantitative analysis

Table 5.3 CHI-SQUARE MATRIX

Actual Sounds / Identified as:

	D	I	R
D	0.015	0.091	0
I	1	0.27	0
R	0	1	0.25

$\chi^2_o = 2.6$

was carried out to confirm this.

The method employed was that of the chi-square test which compares the difference between expected and observed results. Expected results were taken as those from the real sounds, since if the simulations had been identical to the real sounds, the same type of result would have been expected. Observed results were taken as those from the simulations.

The value of χ_o^2 is obtained from

$$\chi_o^2 = \frac{(O - E)^2}{E}$$

where O = observed result

E = expected result

Calculated values of each element in the χ^2 matrix are shown in table 5.3. Summing all the elements in the matrix results in the value of $\chi_o^2 = 2.6$.

The chi-square distribution is related to the normal distribution and depends on the parameter called the 'number of degrees of freedom' (ν). This can be found from,

$$\nu = \text{number of classes} - \text{number of restrictions}$$

Since there are nine pairs of observed and expected results, there are nine classes and since each row of the table must sum to eighty, there is a restriction on the value that can be put in one element of each row making the number of restrictions three in all. Hence the number of degrees of freedom equals six.

Significance tests are usually made at the 0.05 percentage point using the null hypothesis that there is no significant difference between the observed and expected results if the calculated value of

χ_o^2 is less than that given by the tables for the appropriate number of degrees of freedom.

In this present case the chi-square tables yielded a value of 12.59 for six degrees of freedom at the 0.05 percentage point ($\chi^2_{6,0.05}$). The calculated value of χ_0^2 was 2.6. This was much smaller than the value of $\chi^2_{6,0.05}$ and hence the conclusion that there was no significant difference between the observed and expected results held. In fact the results were so similar that there was no significant difference at the 0.80 percentage point ($\chi^2_{6,0.80} = 3.07$).

Hence even although information had inevitably been lost when the percussion sound pressure waveshape was simplified into the four parameter model, sufficient information had been extracted from the original for the reconstructed sounds still to retain the distinctive features of the real sound on both an audible analysis and on a visual inspection of the waveshape, such that the various sounds were still both easily distinguishable and recognisably similar to their originals.

Therefore the four parameter model was considered to be an adequate representation of the percussion sounds; these parameters allowing a quantitative description of the pattern of the various waveforms, and hence rendering the sounds suitable for measurement.

CHAPTER 6

COMPUTER ANALYSIS OF PERCUSSION SOUNDS

6.1 ADVANTAGES

Measuring the four parameters from percussion sounds became an extremely tedious and lengthy task when a considerable number of waveshapes was to be examined. To provide an easier alternative to this laborious technique was the primary incentive for using a computer to automate the measurements and calculation of the four parameters. The computer available was a PDP-12 with an 8k store. Using a computer has two further advantages: the repeatability of measurements, and the increased speed of analysis.

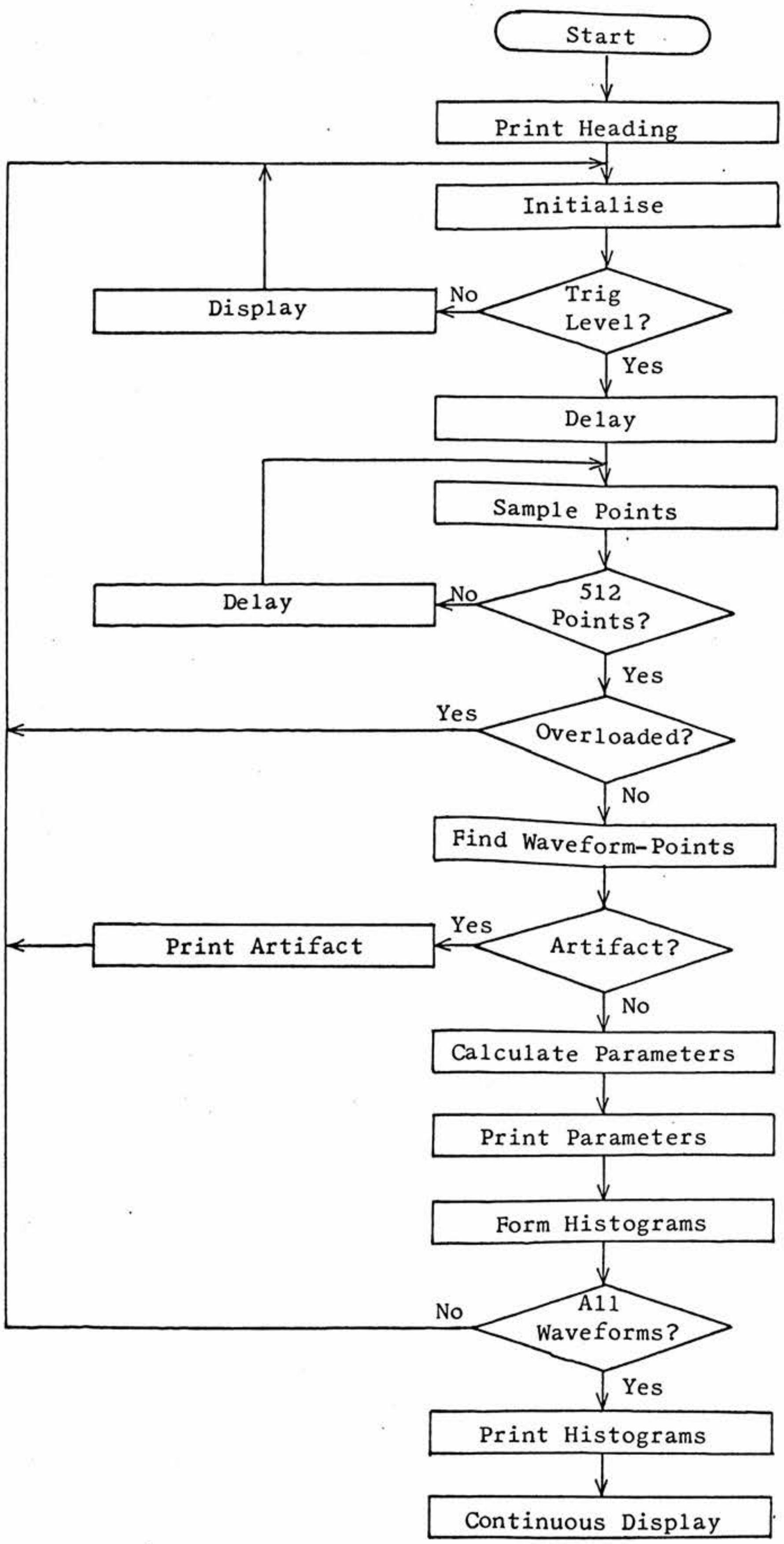
Percussion sounds to be analysed had first to be recorded on magnetic tape, whether the analysis was to be manual or by computer. Then the two methods diverged. The manual method involved transfeering the waveforms on to chart paper. Measurements were then made on the chart paper and the parameters calculated. The measurement of T2 was particularly tedious as it required measuring gradients. This method was employed in the calculation of parameters for simulating the model.

In the new method, the tape was replayed directly to the computer via its analogue-to-digital converters. No further intervention was necessary and the computer printed out the values of the four parameters on the teletype.

Repeatability, was also of importance. Since the four parameters model was only an approximation to any percussion waveshape, the measurement of parameters could vary slightly from one experimenter to another; or even from time to time for the same experimenter. In particular, difficulty could arise in ascertaining the best baseline from which to measure p_o, or in deciding on the best slope for grad B (see figure 5.4). However, when measuring the parameters by computer, exactly the same rules were applied each time. A consistency unattainable by a human analyst was therefore achieved.

Finally, an advantage of speed was also gained. With the laborious chart paper method only about twenty waveforms per hour could be analysed, and after about two hours of analysing even this figure quickly dropped. When using the computer, the limiting factor was not the computer itself but the waiting time between the arrival of each waveform from the tape recorder. Two replay speeds gave the possibility of two rates of analysis. Allowing for 30 s between each sound, replaying at real time would have resulted in an analyses rate of 7200/hour and at one sixtieth real time, 120/hour. The latter rate, although not the maximum, was selected since it allowed monitoring to be carried out while the program was running. After each waveform was analysed and its parameters typed out, the waveform would be displayed on the computer screen until the next waveform arrived. Hence a comparison was able to be made between the typed parameters and the waveshapes themselves. In the debugging of the program this was of immense value, but even after the program was operational it enabled a check on the operation of the program to be maintained. This was to ensure that the entire range of percussion sounds were still capable of being handled by the program. Although the full potential speed of the computer was not used, the increase in speed was found to be adequate, especially as the analysis rate could be kept up indefinitely - an impossible task for the human.

Figure 6.1 FLOW DIAGRAM



6.2 THE PROGRAM

6.2.1 Program Languages

Consideration was first given to the use of a high level language in which to write the program. Although several of these languages were available, none was sufficiently flexible to enable the analysis of an analogue percussion waveform to be attempted. FOCAL-12 was the only available one which allowed the analogue-to-digital converter to be addressed, but amongst other problems, access to the stored samples was not possible. Hence the program had to be written in symbolic language.

Both PDP-8 and LINC were permissible symbolic languages on the PDP-12 computer, and as each had its own particular advantages, the program alternated between the two.

6.2.2 Overall Program

A flow diagram of the final program is shown in figure 6.1.

After printing a heading for each column of parameters, a search was made by one of the analogue-to-digital converters for a percussion waveform on the 'advance' channel of the tape recorder. When one was found another converter sampled, after the necessary delay, the oncoming waveform. Since a complete display on the cathode ray tube requires 512 samples, 512 were taken and stored, that being more than ample to allow the measurements to be made. On real time a delay of only 32 μs existed between each sample, allowing, by the Sampling Theorem a reconstruction of all frequency components from 0 Hz up to over 15 kHz. The actual sampling rate was 520 samples/s.

These samples were checked for any overload before a search was made through them for various points on the waveform. Specific features were searched for and the peaks and troughs of those features identified.

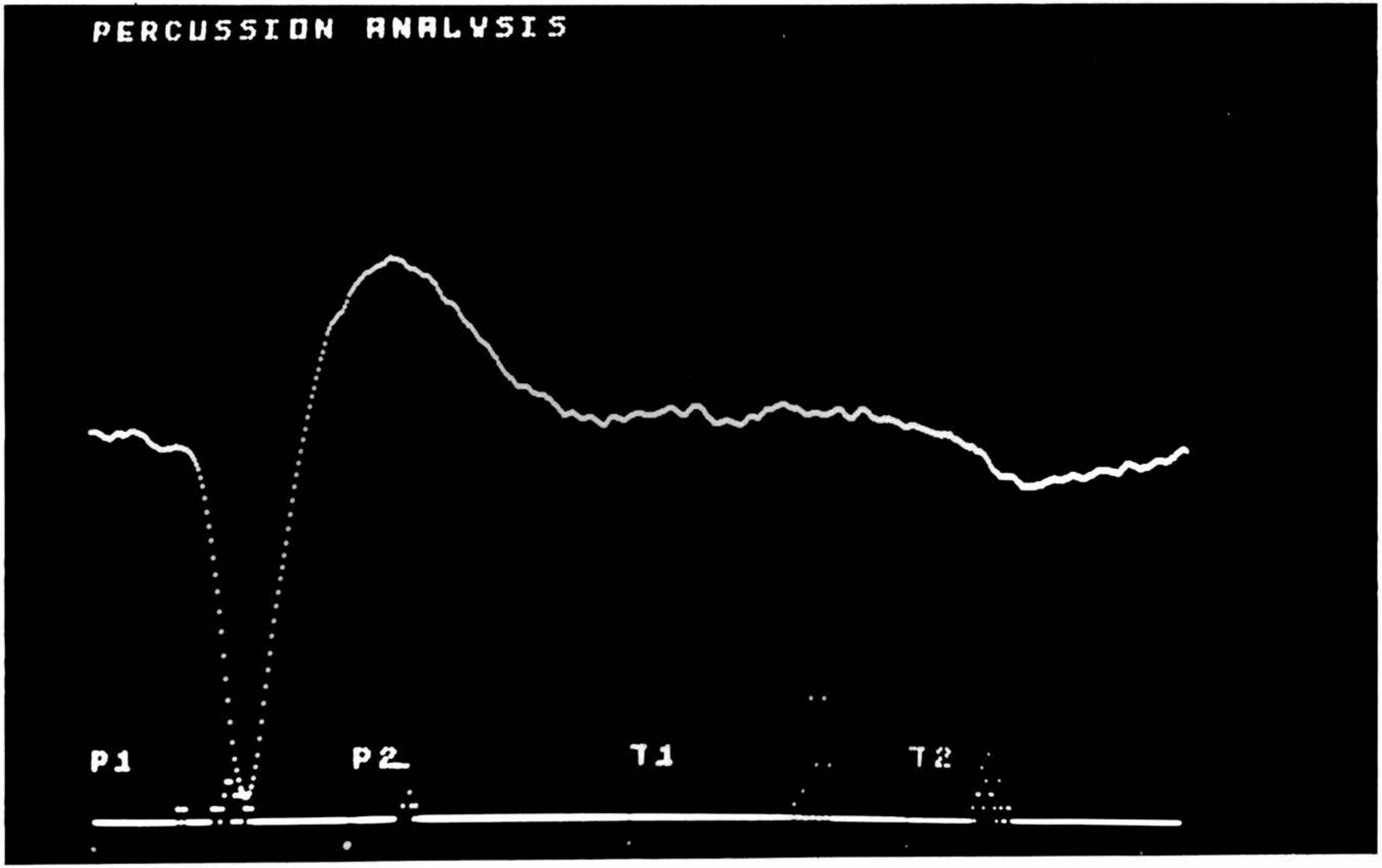
Figure 6.2 COMPUTER PRINT-OUT OF PARAMETERS

NO.	P1	P2	T1	T2
0001	0073	0033	0108	0036
0002	0076	0036	0102	0037
0003	0076	0038	0105	0039
0004	0075	0035	0108	0042
0005	0069	0033	0121	0042
0006	0065	0031	0128	0040
0007	0074	0034	0124	0042
0008	0064	0032	0124	0052
0009	0066	0028	0128	0045
0010	0078	0037	0115	0046
0011	0079	0035	0112	0043
0012	0078	0033	0112	0035
0013	0085	0041	0105	0038
0014	0080	0034	0108	0039
0015	0083	0037	0118	0034
0016	0079	0034	0118	0041
0017	0091	0042	0121	0036
0018	0069	0041	0112	0045
0019	0065	0036	0112	0041
0020	0070	0040	0115	0044

Units:

Parameter	Multiplication Constant
P1	10^{-2}
P2	10^{-2}
T1	10^{-5}s
T2	10^{-4}s

Figure 6.3 COMPUTER DISPLAY AFTER WAVEFORM ANALYSIS



Next, a simple check was carried out to ascertain if the waveform was likely to be an artifact. If so, 'artifact' was typed out by the teletype, and the program returned to find the next waveform.

From the points identified, the four parameters were calculated and scaled before being typed out in the columns already headed. Figure 6.2 illustrates a typical computer output after the analysis of twenty waveforms. The table appended contains the parameter units.

After the parameters had been typed, the program returned to search for the next waveform. Simultaneously with the search, the previous waveform was displayed, enabling the parameters just calculated to be compared with the waveform on the computer screen.

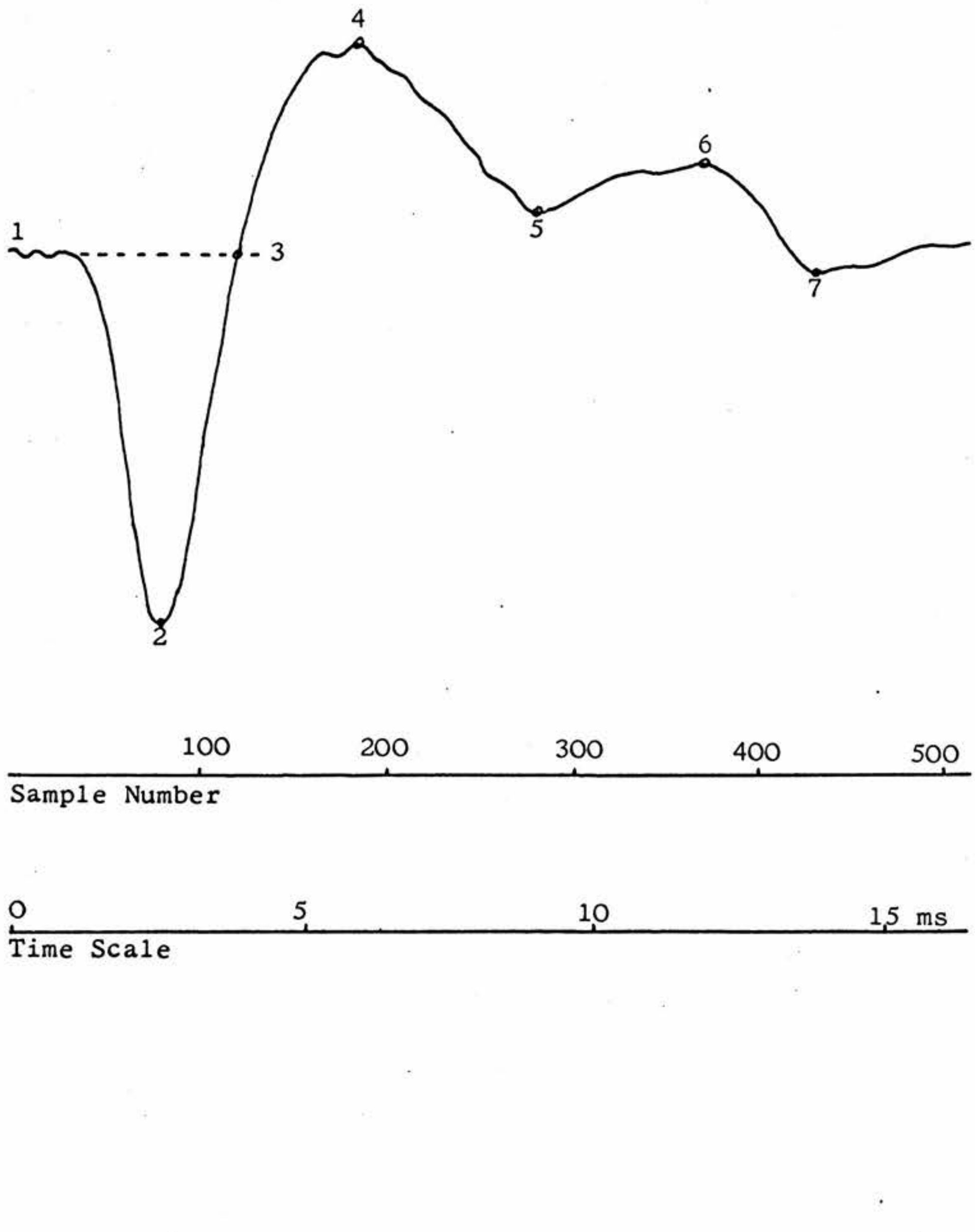
The program could either be left running continuously in this mode or be set to accept a specific number of waveforms. Then histograms, tabulating the frequency of occurrence of all values of each parameter would be typed out. During the display routine these histograms were also displayed.

Finally the program would enter a continuous display loop and remain there until the computer was reset to analyse the next batch of waveforms. Figure 6.3 illustrates a typical computer display of a percussion waveform along with the four histograms formed. As the histogram displays were only required to give some idea of the variation in parameters while the analysis was proceeding, no values were attached. The exact values were given in the print-out if required.

6.2.3 Overload Detection

The overload detection routine tested each sample to check if the input had reached, and hence could have exceeded, the acceptable limits of the analogue-to-digital converter at any time during the sampling.

Figure 6.4 WAVEFORM-POINTS SEARCHED FOR BY COMPUTER



Most overloads occurred on the rarefaction peak, and as this would alter the value of p_o, errors in the calculation of the parameters P1 and P2 would then result. In addition, T1 could not be measured accurately because of the flat cut-off region on the rarefaction. Hence the calculation of T2 was also affected. Therefore any waveform which overloaded the analogue-to-digital converter was discarded, and the program returned to search for another waveform.

'Drop-outs' on the magnetic tape were also detected since they always produced overload voltages. Hence no attempt was made to do any calculations if a 'drop-out' occurred on the waveform.

6.2.4 Finding Points on Waveform

Seven separate points were searched for on each percussion sound waveform to be analysed. These points are numbered on a typical trace shown in figure 6.4.

By taking the average of the first eight samples of the trace, the baseline or point 1 was found. Averaging was used to reduce the effect of noise on the trace. This point established the baseline from which p_o would be measured (see figure 5.4).

Proceeding through the samples sequentially, the tip of the rarefaction spike was identified by the most negative sample and its value stored. At the same time the sample number was also stored to enable the time parameters to be calculated. The other point required in the calculation of T1 was point 3. This was the position at which the waveform had just crossed the baseline, and was found by comparing these samples after the rarefaction peak with point 1.

Point 4, the following maximum compression peak was found by comparing every sample with the maximum previously encountered and updating it by replacing it if a new maximum was found. The program jumped out of this 'find maximum' subroutine only after the sample value fell to a fixed level below the maximum or when the 320th sample was reached. By allowing the search to be made over all of the top of the first compression peak, this prevented latching on to the wrong peak of a noisy waveform.

Emerging from that loop, the program entered another loop to search for points 5 and 6. They were found by a method similar to that for point 4. In addition every time a new minimum was found the value of point 6 was reset to zero so that only that maximum encountered after point 5 was actually stored. After the 320th sample was passed the value of point 5 was stored and the next maximum (point 6) and minimum (point 7) were found.

6.2.5 Parameter Calculations

Values of P1 and P2 were calculated simply by subtraction and a 'divide' routine.

T1 was calculated by converting the number of samples between points 2 and 3 into a time by multiplying by 32 μs. No attempt was made to interpolate between samples or to reduce the delay between samples at this stage in order to increase the accuracy in the calculations. In any case it was found to be unnecessary due to the very large variation in T1 and T2 under normal percussion (see Chapter 7).

Before T2 could be calculated, grad A and grad B had to be found. Grad A was found simply by measuring the slope between two points on the positive gradient of the rarefaction. The measurement of grad B presented difficulties due to its noisy and often rather indistinct gradient. The technique which yielded a solution involved measuring the area contained under the slope for a section 32 samples wide, and then computing the equivalent straight line gradient of that section of the slope. Moving one sample to the right each time, this procedure was continued down the negative gradient of the first compression peak. The maximum value was retained as the gradient of the slope. T2 was calculated from formula 5-1.

$$T2 = T1 \; \frac{\text{grad A}}{\text{grad B}}$$

6.2.6 Artifact Detection

To prevent the computer attempting to analyse an artifact, a simple test was carried out to check for the existence of the rarefaction, since that was the most prominent feature of the waveform. Artifacts were caused by such phenomena as large rarefaction pressure changes due to wind or the closing of doors during the recording session, or by a 'drop-out' on the 'advance' channel resulting in sampling when no waveform existed.

Less than 5% of waveforms sampled were artifacts, but this figure was considered troublesome enough to warrant artifact detection.

The simple test, which only tested for a particular minimum threshold on the rarefaction spike, although not identifying every artifact, proved to work sufficiently well. Out of a batch of over seven hundred sounds analysed, an attempt had been made to analyse less than 20% of artifacts contained within the batch. The other artifacts had been correctly rejected as such.

CHAPTER 7

RESULTS FROM THE FOUR PARAMETER MEASUREMENTS

In this chapter, the quantitative analysis of percussion sounds from healthy subjects and from subjects with a pathological condition in one lung will be presented.

The intention, while carrying out the work, was not to provide a complete analysis of all possible sounds. Instead, a number of different sounds were analysed, and with these examples it is hoped to demonstrate the possibilities of four parameter analysis.

With automation of parameter measurement and calculation, it was now possible to proceed with the analysis of a large number of waveforms. Due to the random variation in waveform caused by fluctuation in some external variable which could not be held exactly constant, such as a changing respiration level or percussion force, it was desirable to analyse a large batch of waveforms for every sound considered.

As random variation in a batch of percussion sounds could not be avoided, it was important to discover if this random variation made it impossible to observe changes in the quality of sound. To test this, the four parameters were measured for cases which were known to produce small changes in sound quality. Two different groups of percussion sounds were analysed, one for changes in lung volume, and the other for changes in percussion position on the chest.

7.1 SIZE OF SAMPLE BATCH

As a first step to considering random parameter variation, the size of sample batch had to be considered. Normally the larger the sample batch, the smoother the frequency distribution curve which will be obtained. Here, a number of factors affects the size of sample batch.

Firstly, there is an upper limit on the number of samples due to the difficulty in retaining a constant lung volume for any length of time. This is particularly so at low lung volumes. When sounds recorded at the residual volume (RV) of the lungs are to be analysed, a maximum of about twenty sounds is all that is possible.

Secondly, there is the problem of inconsistent percussion blows. Two important factors had to be considered. One would have had an opposite effect to the other on the choice of the number of samples. A large number of samples might have resulted in an increase in the randomness of results due to the investigator becoming tired and hence reducing the consistency of his percussion blows. This would have necessitated a small number of samples. On the other hand, the first few samples of a batch might have been less consistent than the ones following if several percussion blows are required before the investigator settles down to a consistent technique. If true, this would have required a large number of samples.

In order to resolve this difficulty, comparisons were made between large and small batches of samples. Five batches of ten samples and one of forty samples of similar 'resonant' percussion sounds were analysed. From each batch, the standard deviations of all four parameters were calculated, and the average standard deviation of the small batches found. Comparing the standard deviation of the small and large batches of samples no significant difference could be found. Hence up to at least forty samples, the randomness of results was not found to be dependent on the size of sample. Nevertheless large numbers of samples were used where possible to obtain a smoother frequency

Figure 7.1 VARIATION IN PARAMETER P1 WITH LUNG VOLUME

Position - 4th i.c.s., right hand side

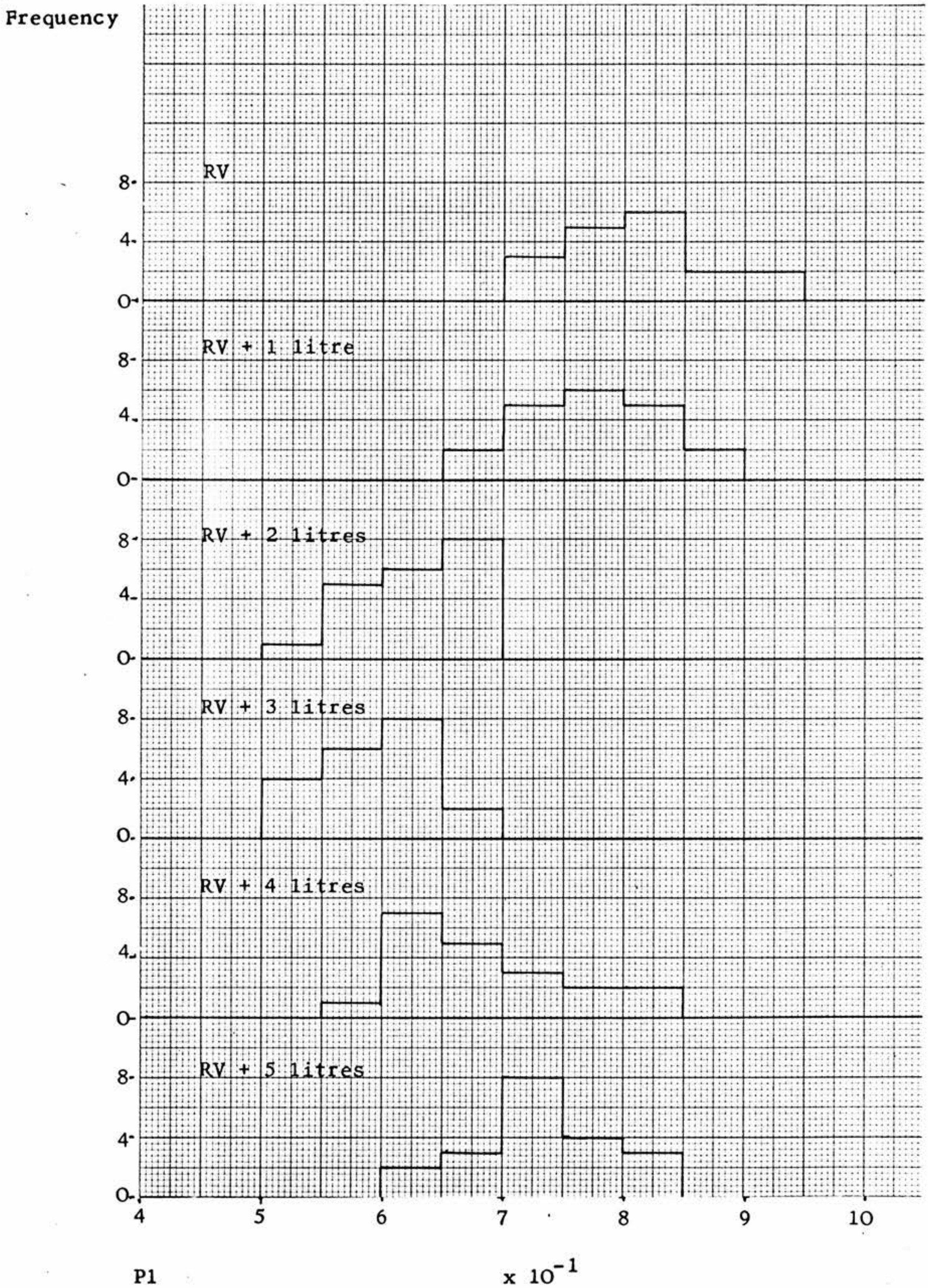


Figure 7.2 VARIATION IN PARAMETER P2 WITH LUNG VOLUME

Position - 4th i.c.s., right hand side.

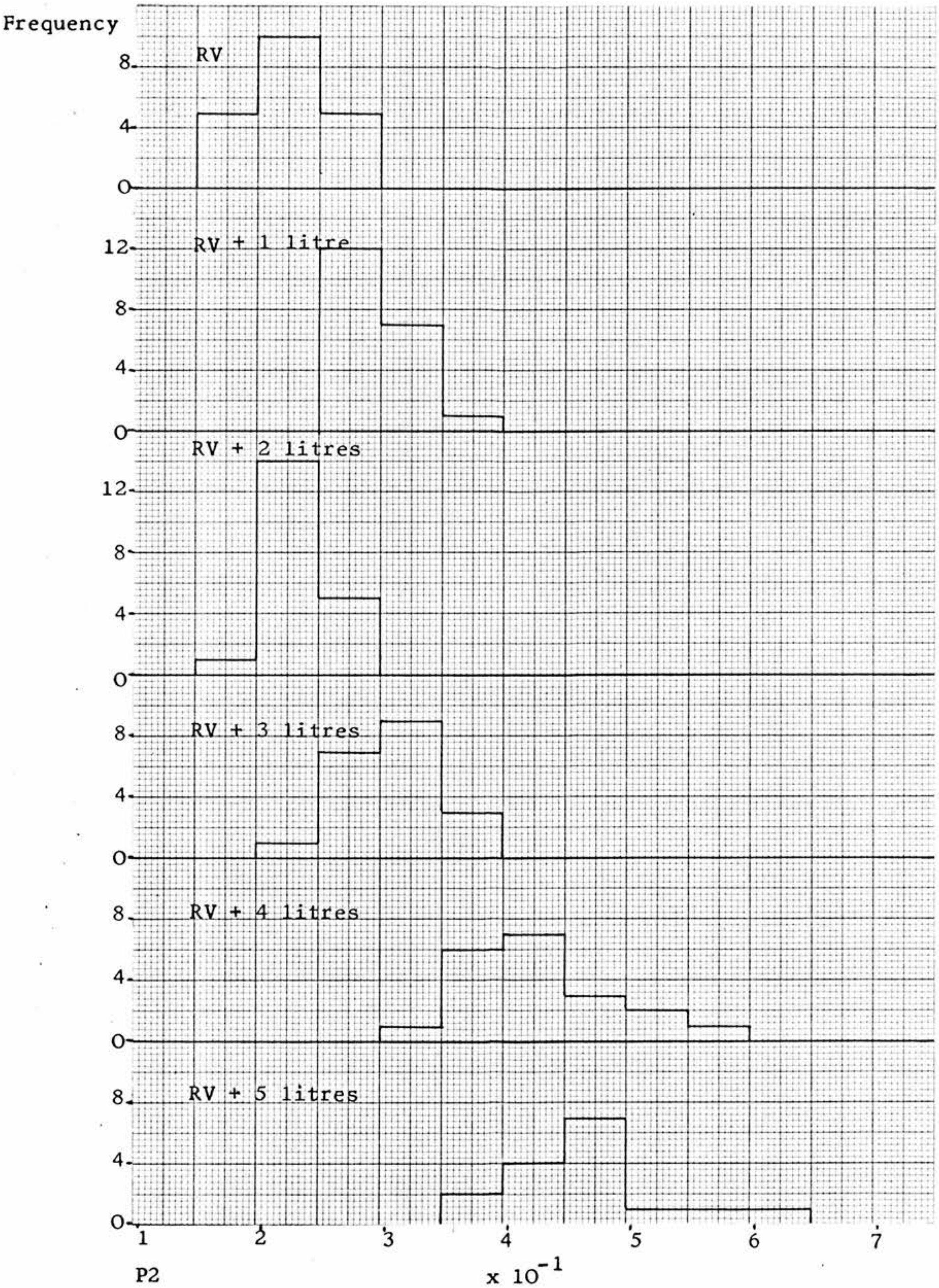


Figure 7.3 VARIATION IN PARAMETER T1 WITH LUNG VOLUME

Position - 4th i.c.s., right hand side.

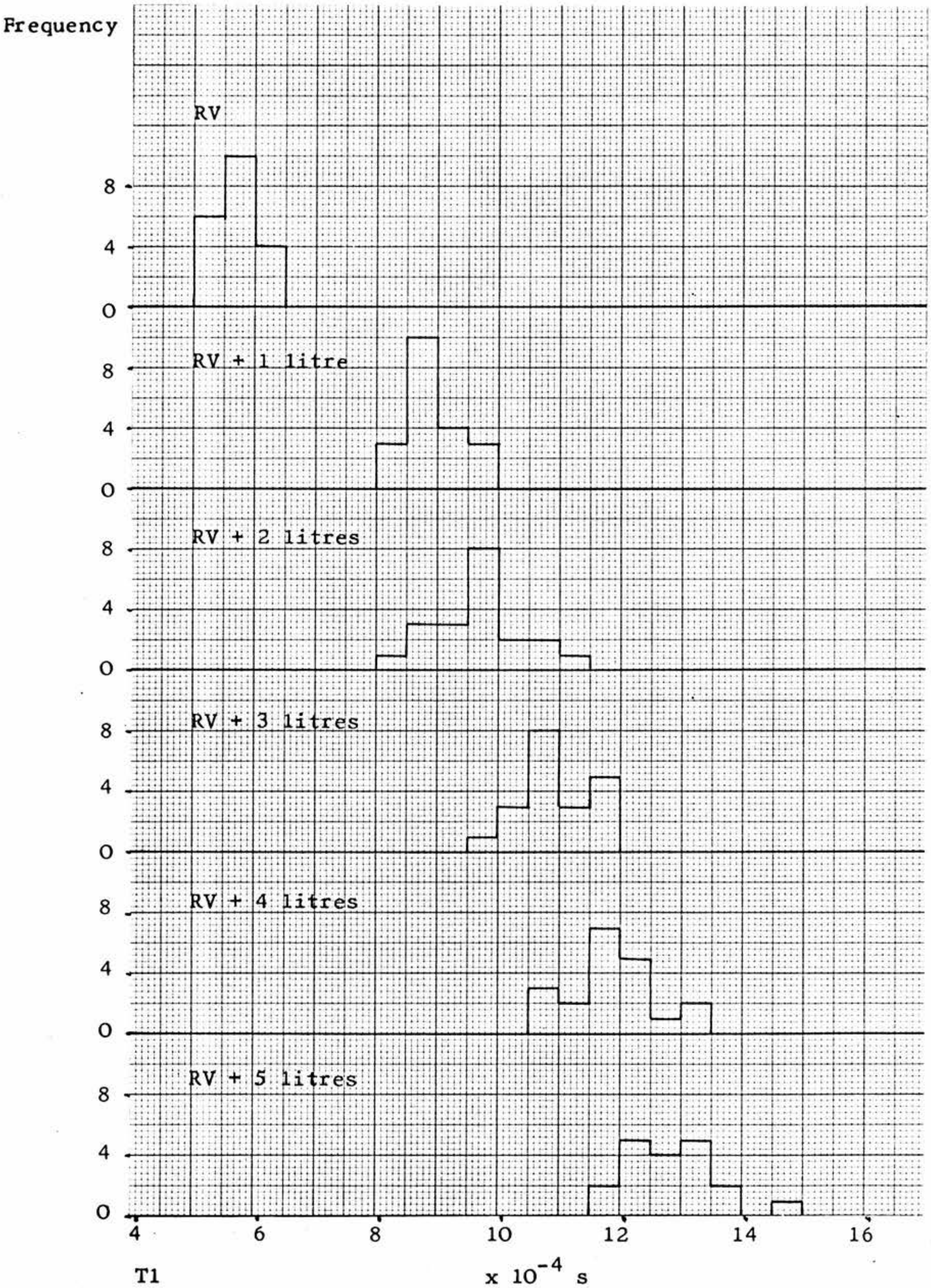
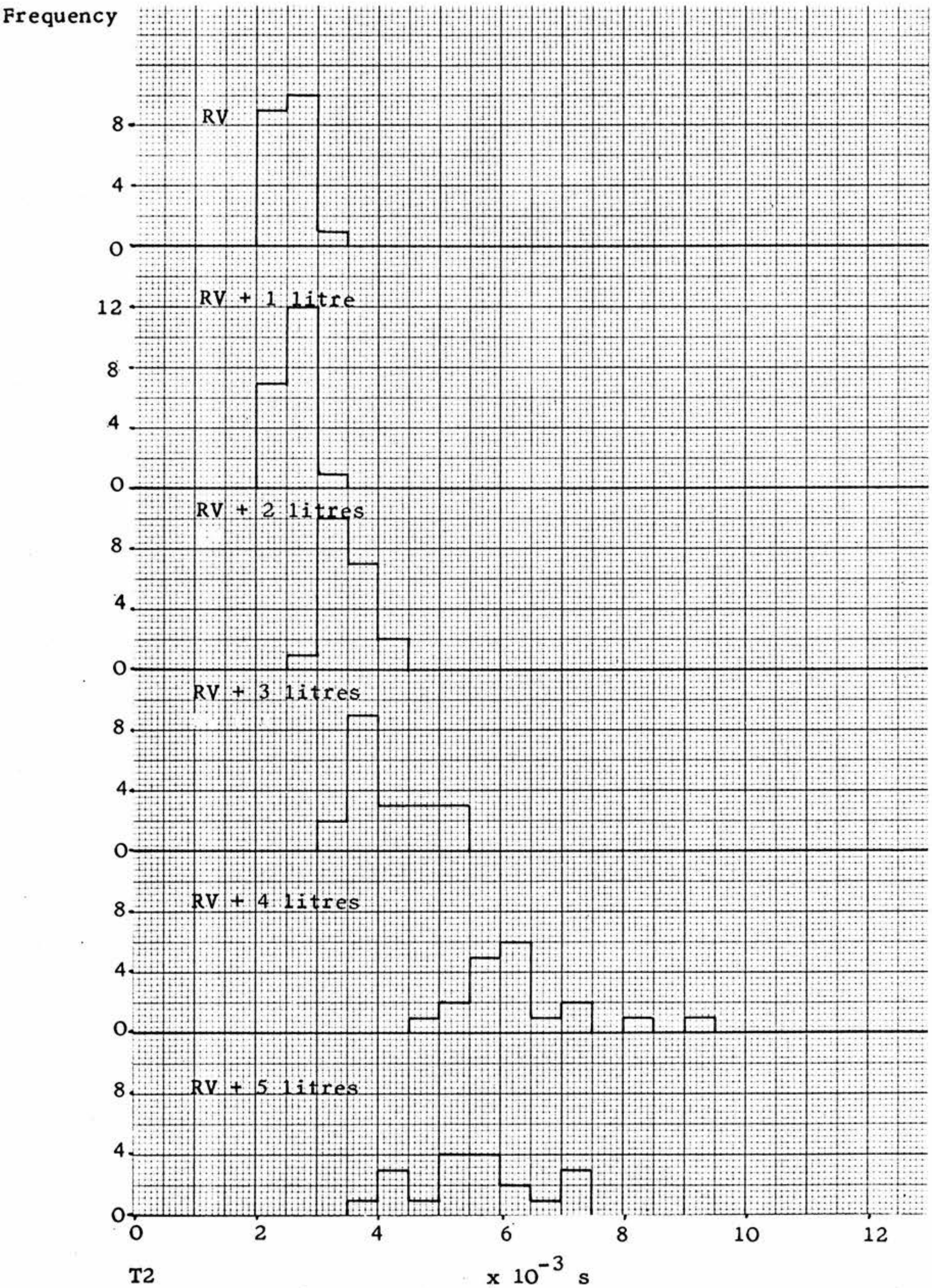


Figure 7.4 VARIATION IN PARAMETER T2 WITH LUNG VOLUME

Position - 4th i.c.s., right hand side.



distribution curve for the parameters considered.

7.2.1 Changes in Lung Volume

Percussion sounds recorded with the lungs at various levels of inspiration were analysed. The lung capacities considered varied from the residual volume (RV) in steps of one litre to the total lung capacity of the subject - in this case, RV + 5 litres. The sounds were taken from the fourth intercostal space on the right-hand side. Such a position ensured that the sound was 'resonant' at all lung volumes.

Since sounds from the lung at the RV were to be analysed, the maximum number of sounds per group had to be limited to twenty.

After the analysis, the frequency distributions of all four parameters were plotted in the histograms of figures 7.1-4. Most of the parameter changes showed definite trends which were observable despite the random variation.

P1, with increasing lung volume, began at first to decrease, but at about RV + 4 litres it started to increase. Note that on the judgement of medical practitioners the 'resonant' quality of the sound rises with inspiration.

P2 and T1 both show a steady increase with increasing lung volume, that of T1 being the more noticeable.

T2 yielded a rather erratic variation when the lung volume rose above RV + 4 litres. At such high lung volumes the size of the second compression peak becomes so large that it tends to obscure the gradient from which T2 is measured (grad B of figure 5.4), making it particularly difficult to evaluate T2.

From the study of the parameters above it can be seen that there are in fact observable trends in the measurements made. The random

Figure 7.5 VARIATION IN PARAMETER P1 WITH POSITION ON CHEST

Lung Volume - FRC

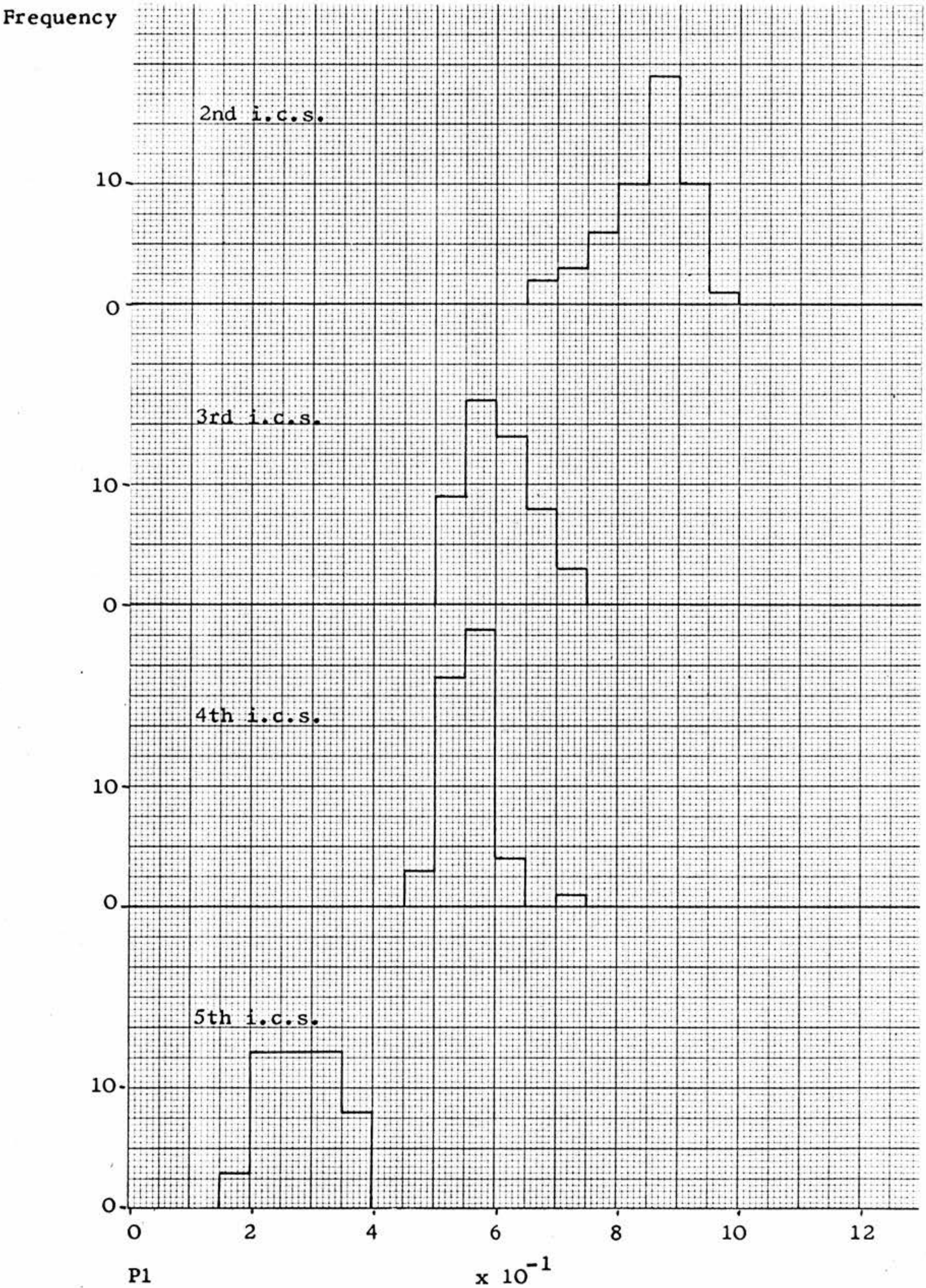


Figure 7.6 VARIATION IN PARAMETER P2 WITH POSITION ON CHEST

Lung Volume - FRC

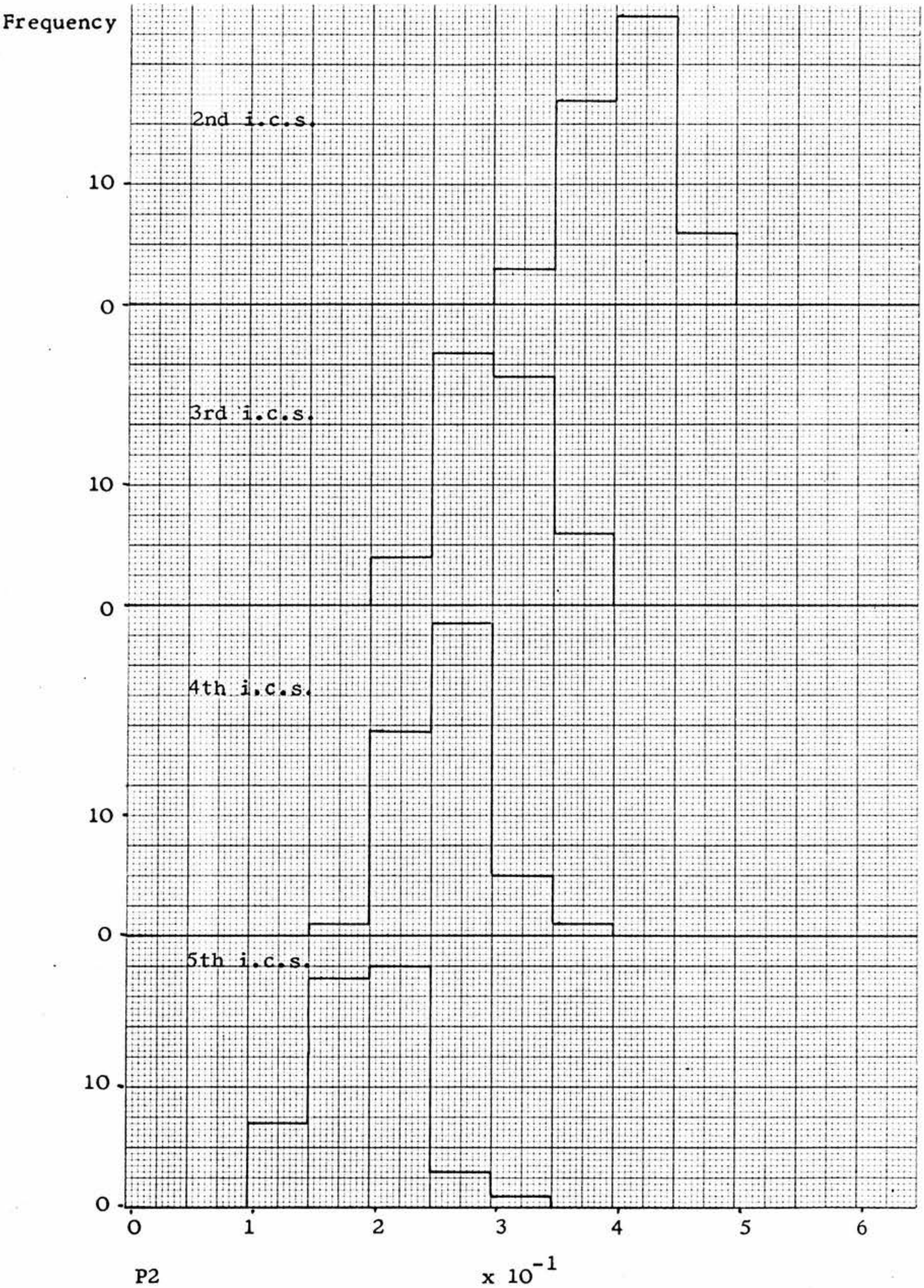


Figure 7.7 VARIATION IN PARAMETER T1 WITH POSITION ON CHEST

Lung Volume - FRC

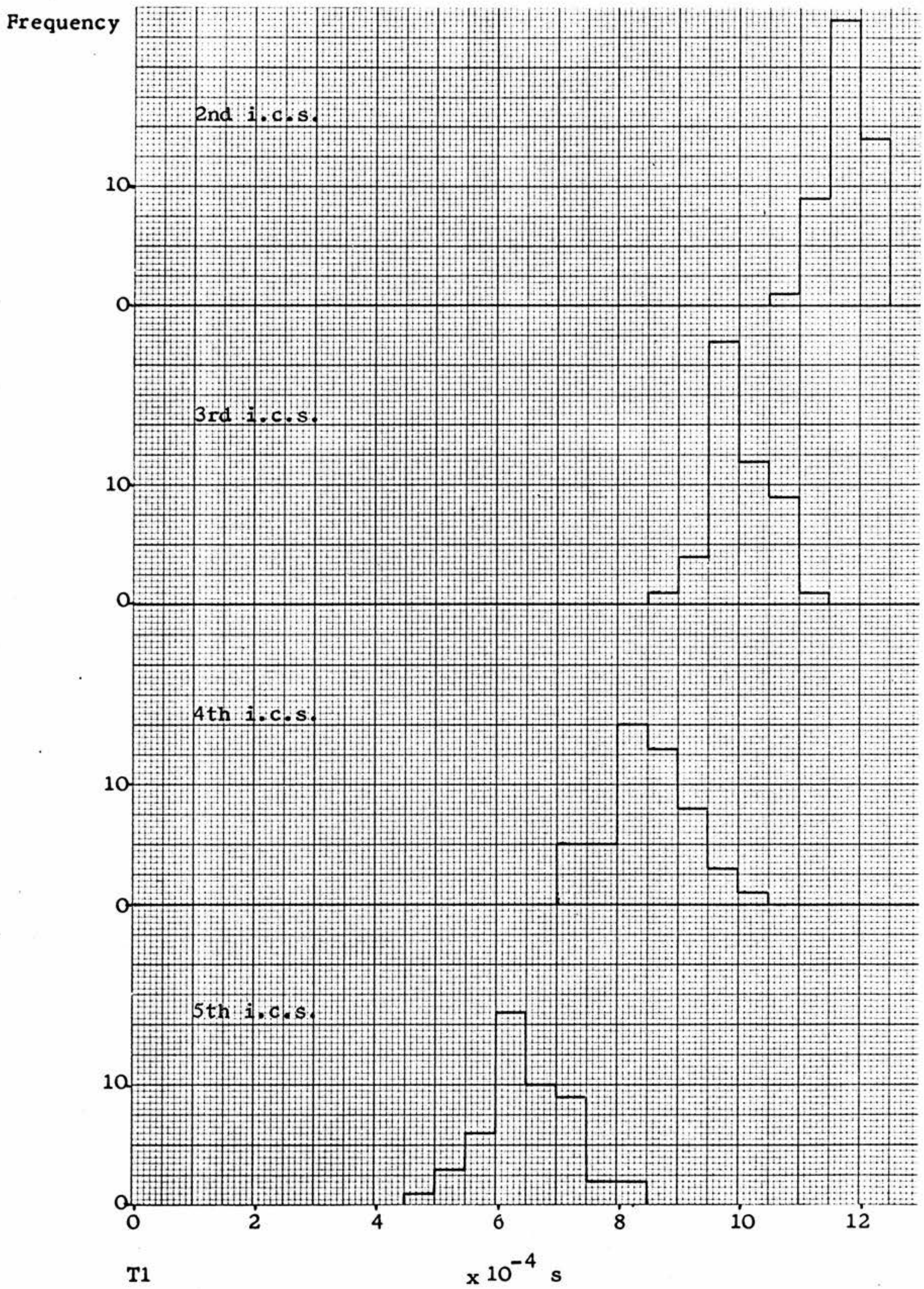
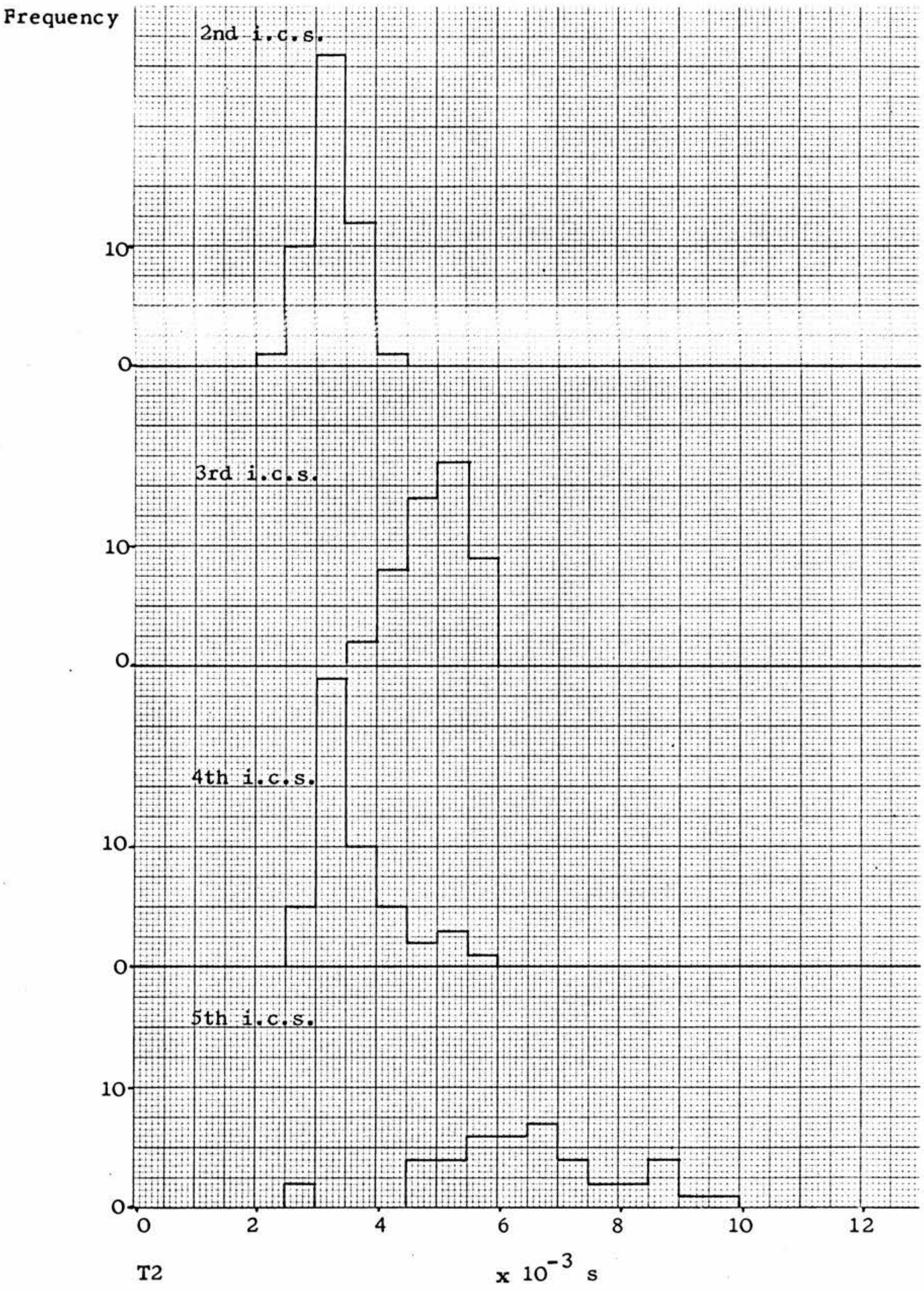


Figure 7.8 VARIATION IN PARAMETER T2 WITH POSITION ON CHEST

Lung Volume - FRC



variations of parameters are not so great that they mask the change in quality of the sounds.

4.2.2 Changes in Percussion Position

Histograms were this time plotted to show the variation in parameters for different positions on the chest. The lung volume was kept at its FRC as this allowed large sample batches to be analysed. Fifty sounds were analysed for each batch.

Figures 7.5-8 show the frequency distribution histograms obtained. Again a distinct trend could be isolated in the parameters P1, P2 and T1. The measurement of T2, however, was not quite so satisfactory this time.

P1 fell with decreasing 'resonance', i.e. with moving the percussion position down the chest. This change was the opposite to that discovered in the previous section where, at normal inspiration levels, an increase in 'resonance' (with increasing lung volume) produced a decrease in P1.

P2 also fell with decreasing 'resonance', although this time the overlap in histograms was greater. Nevertheless it can be seen that the separation between sounds from the second and fourth, and from the third and fifth intercostal spaces was well defined.

T1 illustrates a very distinct change with position on chest.

T2 turned out to be rather random with no noticeable trend. Large variations in parameter value at the fifth intercostal space can be accounted for because of the small values of P1 (small first compression peak) and hence a rather indistinct gradient from which T2 is measured.

Again it can be concluded that the trend in parameter change was generally greater than the random variation. Hence in practice the four parameter model proved to be satisfactorily suited to discriminating

Figure 7.9 P1 - T1 CLUSTER DIAGRAM FOR VARIOUS LUNG VOLUMES

Position - 4th i.c.s., r.h.s.; 20 samples per group; > 80% of samples enclosed.

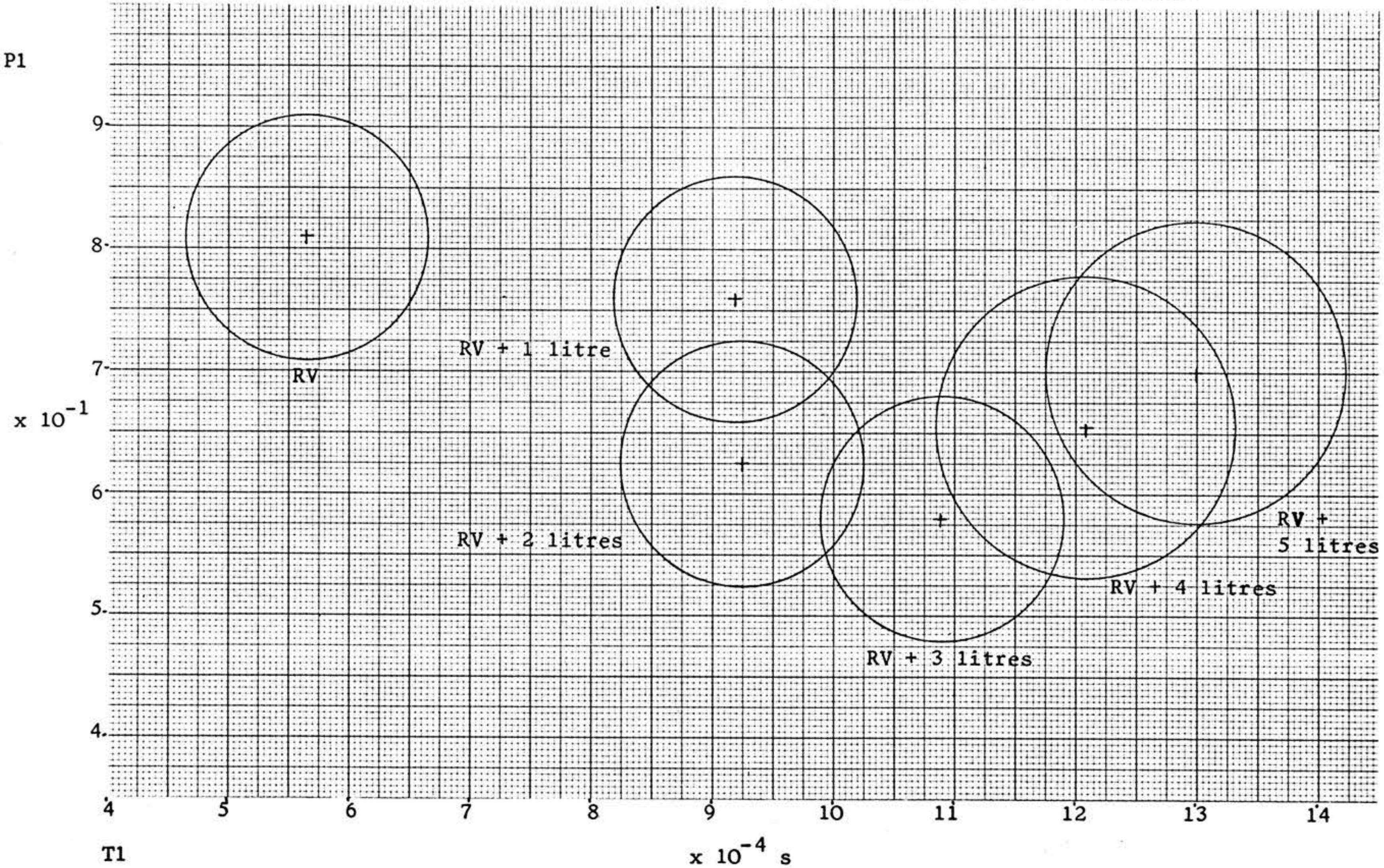


Figure 7.10 T2 - T1 CLUSTER DIAGRAM FOR VARIOUS LUNG VOLUMES

Position - 4th i.c.s., r.h.s.; 20 samples per group; > 80% of samples enclosed

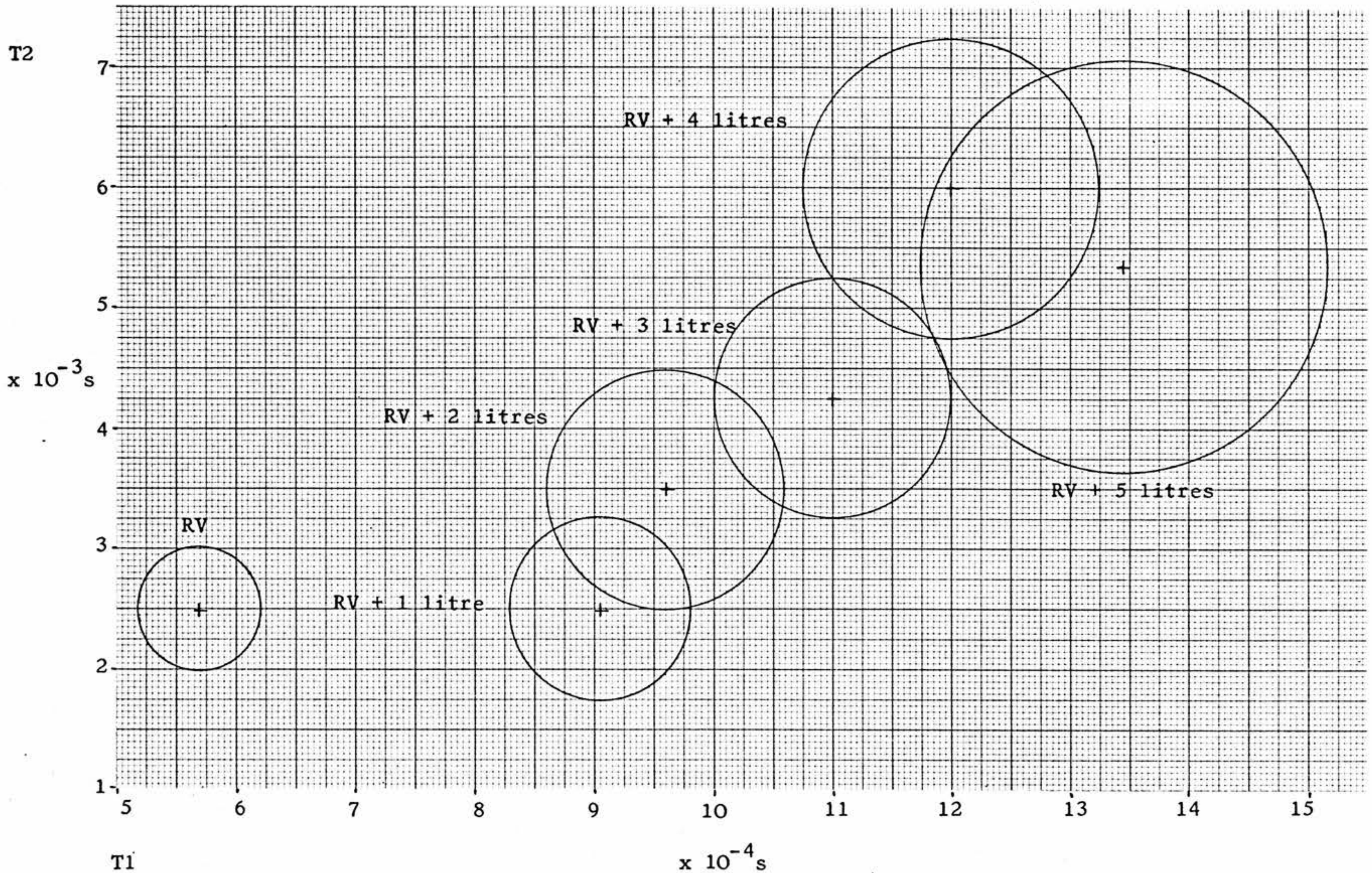
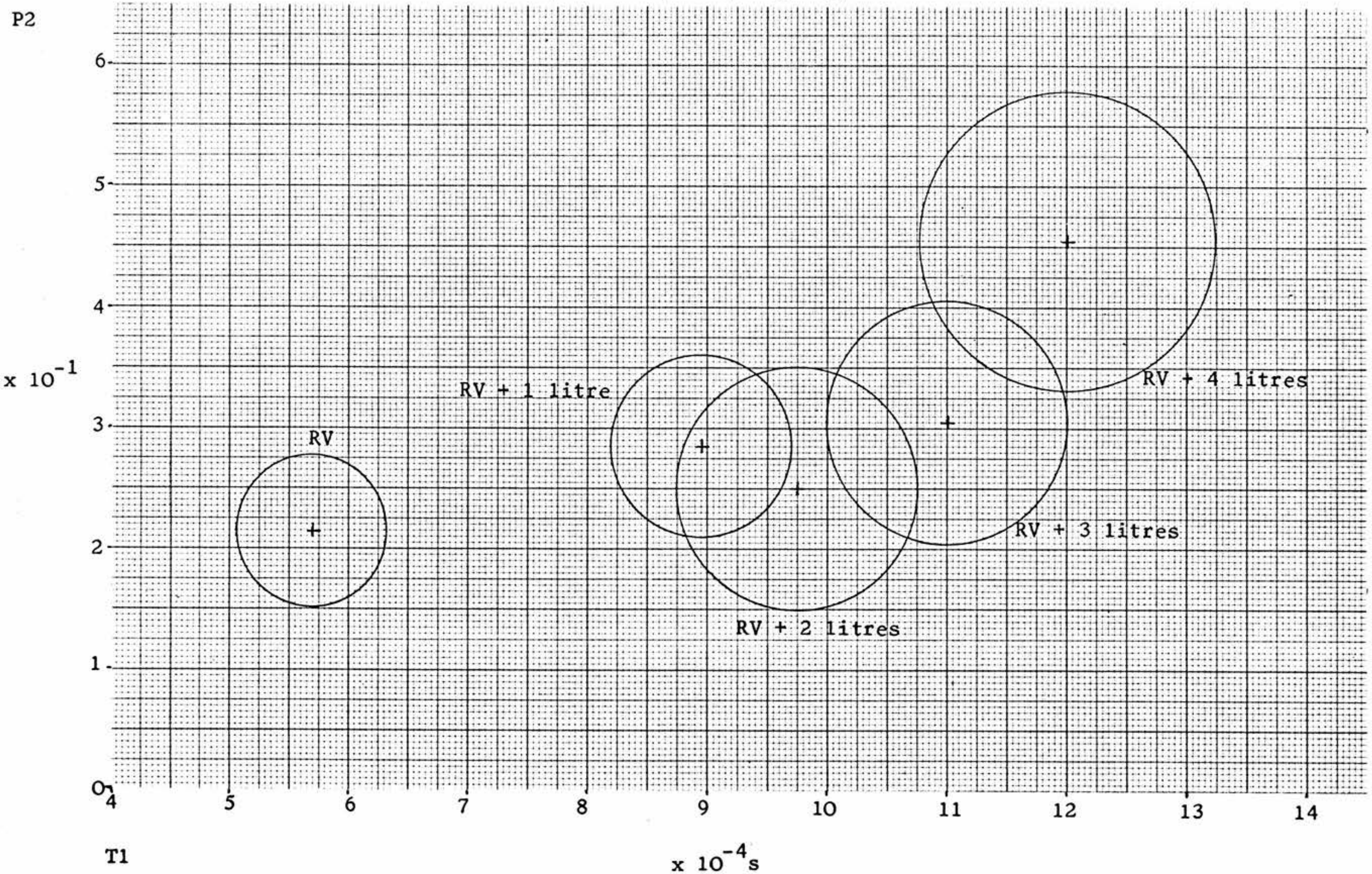


Figure 7.11 P2 - T1 CLUSTER DIAGRAM FOR VARIOUS LUNG VOLUMES

Position - 4th i.c.s., r.h.s.; 20 samples per group; > 80% of samples enclosed



between various 'resonant' percussion sounds.

7.3 CLUSTER DIAGRAMS

Histograms enable comparison of only one parameter at a time. This is a one dimensional analysis. Each sound, however, has four parameters and hence to map every sound fully requires a four dimensional space. A single diagram cannot be used to display four dimensions, although it is possible to display three using an isometric view. Such a diagram is not altogether satisfactory due to the difficulty in visualising where points lie in three dimensions.

In this present work only two dimensional cluster diagrams will be used, and where necessary a number of these cluster diagrams will be shown so that the relationship between all four dimensions can be deduced. Generally, however, only those diagrams which best illustrate the features under discussion will be given.

To show where points are distributed within a hyperspace, a hypersurface can be constructed to enclose all points, or a certain proportion of points, of similar type. In three dimensions the hypersurface reduces to an ordinary surface, and in two dimensions, to an enclosing curve.

Cluster diagrams showing the relationship between parameters for each group of sounds at different lung volumes have been drawn in figures 7.9-11. The data used were from those sounds analysed for the histograms of figures 7.1-4.

The three cluster diagrams yield the information required for constructing the hypersurfaces to enclose each group of points. Circles were used on the diagrams in order to imply no special distribution for the points, since the measurements provided no evidence for any special

Figure 7.12 P1 - T1 CLUSTER DIAGRAM FOR VARIOUS POSITIONS ON CHEST

Lung volume - FRC; 50 samples per group; > 80% of samples enclosed

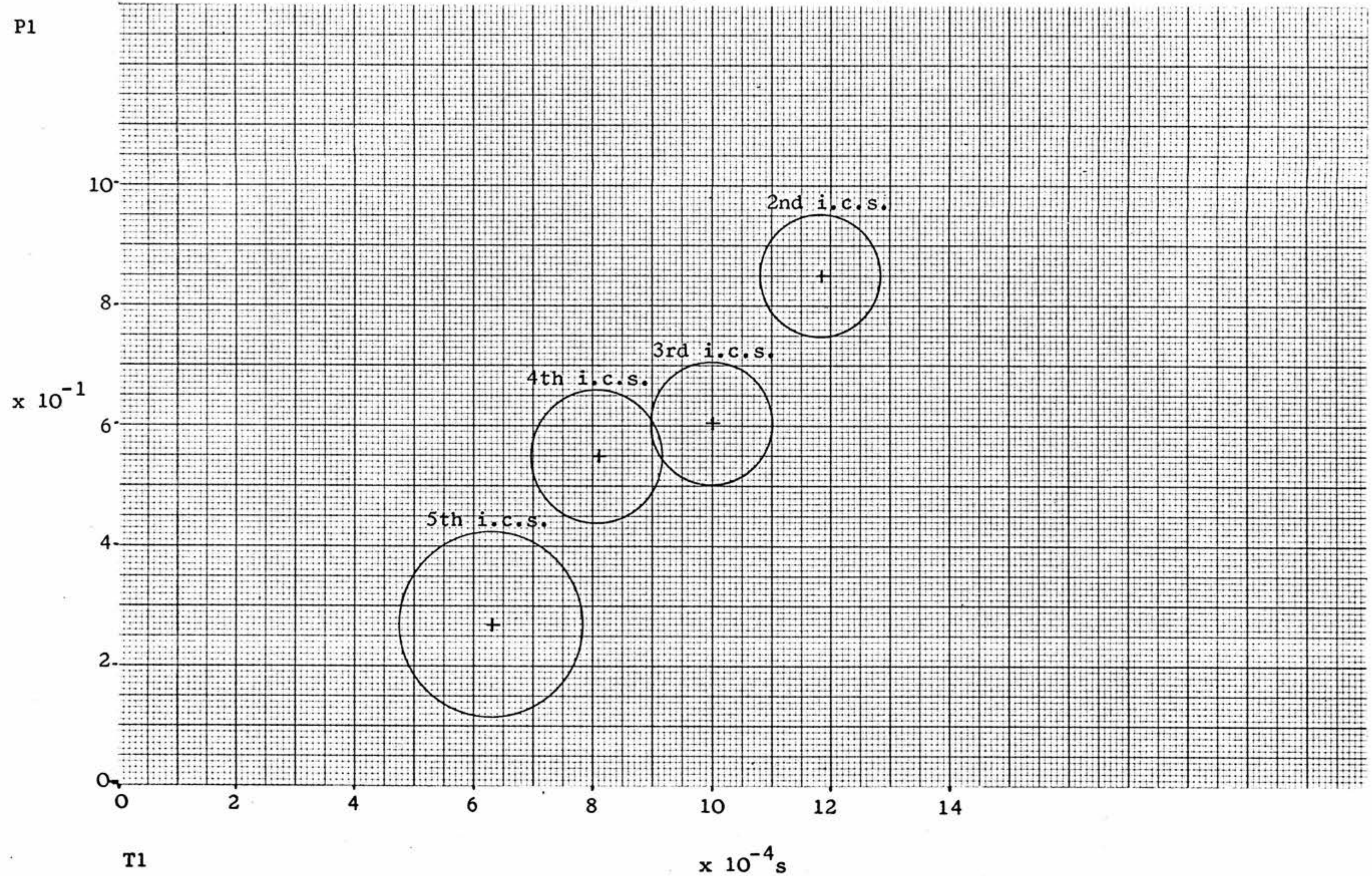
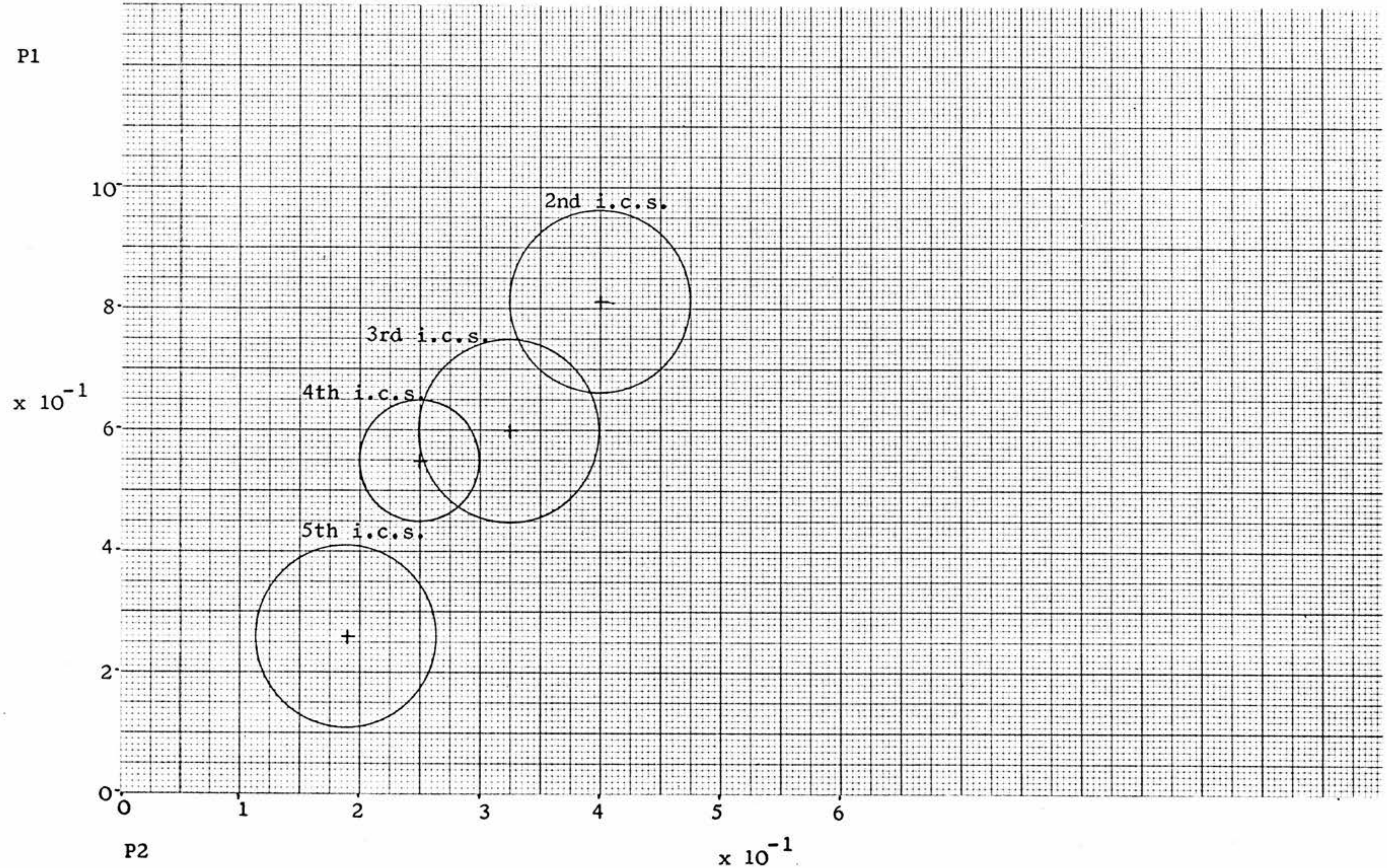


Figure 7.13 P1 – P2 CLUSTER DIAGRAM FOR VARIOUS POSITIONS ON CHEST

Lung volume - FRC; 50 samples per group; > 80% of samples enclosed



distribution; most of the points being reasonably uniformly distributed around the centre of the circle.

Another set of cluster diagrams (figures 7.12-13) illustrates the effect of percussion position on the sounds. The sounds analysed for the histograms of figures 7.5-8 provide the data for the cluster diagrams.

While the original histograms showed that there was often considerable overlap in the parameters of different groups of sounds, cluster diagrams separate the sounds out in two dimensions and result in a far greater discrimination between sounds. If two cluster diagrams are visualised together to form a pattern in three dimensional space, the clusters separate out further still.

7.4 CLUSTER ANALYSIS

Clusters can be easily located and their size given simply by quoting the values of the appropriate parameters. However it must be remembered that physicians note differences between sounds, and so it was decided to measure the separation between clusters of sounds.

Cluster separation can be measured in two ways. Either the separation on all four parameters can be obtained separately or a single measurement can be made in four dimensional space.

Knowing the difference between sounds in all four parameters retains the greatest amount of information. However, this makes a comparison of the difference difficult. A single figure measurement is best for such a purpose.

It was, therefore, decided to measure the spatial separation of clusters. Before this could be attempted, the scales and units of the four parameters had to be standardised in some way. P1 and P2 were already dimensionless, and so to obtain similar units for T1 and T2 they had to be divided by figures whose units were in seconds.

Often in an analysis of this type, the standard deviations of the values of each parameter are used for normalising. This prevents the randomness of any one parameter having a disproportionate effect on the randomness of the separation between a series of two samples. Generally, such a method is restricted to a study where the scales are to be weighted to enable results already obtained to be analysed. For example, in a sociological study information (e.g. IQ, personality) can be obtained for different races to see if there is any significant difference between the races. In this present study, on the other hand, an attempt had to be made to weight the scales of the four parameters in such a way that this weighting could be used in the analysis of all possible percussion sounds. Because of the vast range of sounds it was not possible to predict a value of standard deviation which could be usefully employed.

Instead, after consideration of the results from sounds already analysed, divisors for each of the parameters were selected.

Parameter	Divisor for weighting parameter scales
P1	0.1
P2	0.05
T1	10^{-4} s
T2	10^{-3} s

That no special accuracy is claimed can be seen from the fact that three of the four divisors used are factors of ten.

In addition to the criterion discussed above, the choice was also designed to allow the value of each parameter after division to fall

Figure 7.14 P1 - T1 CLUSTER DIAGRAM FOR DISCRIMINATION OF TWO SOUNDS FROM HEALTHY SUBJECTS

Lung volume - FRC; 20 samples per group; > 90% of samples enclosed

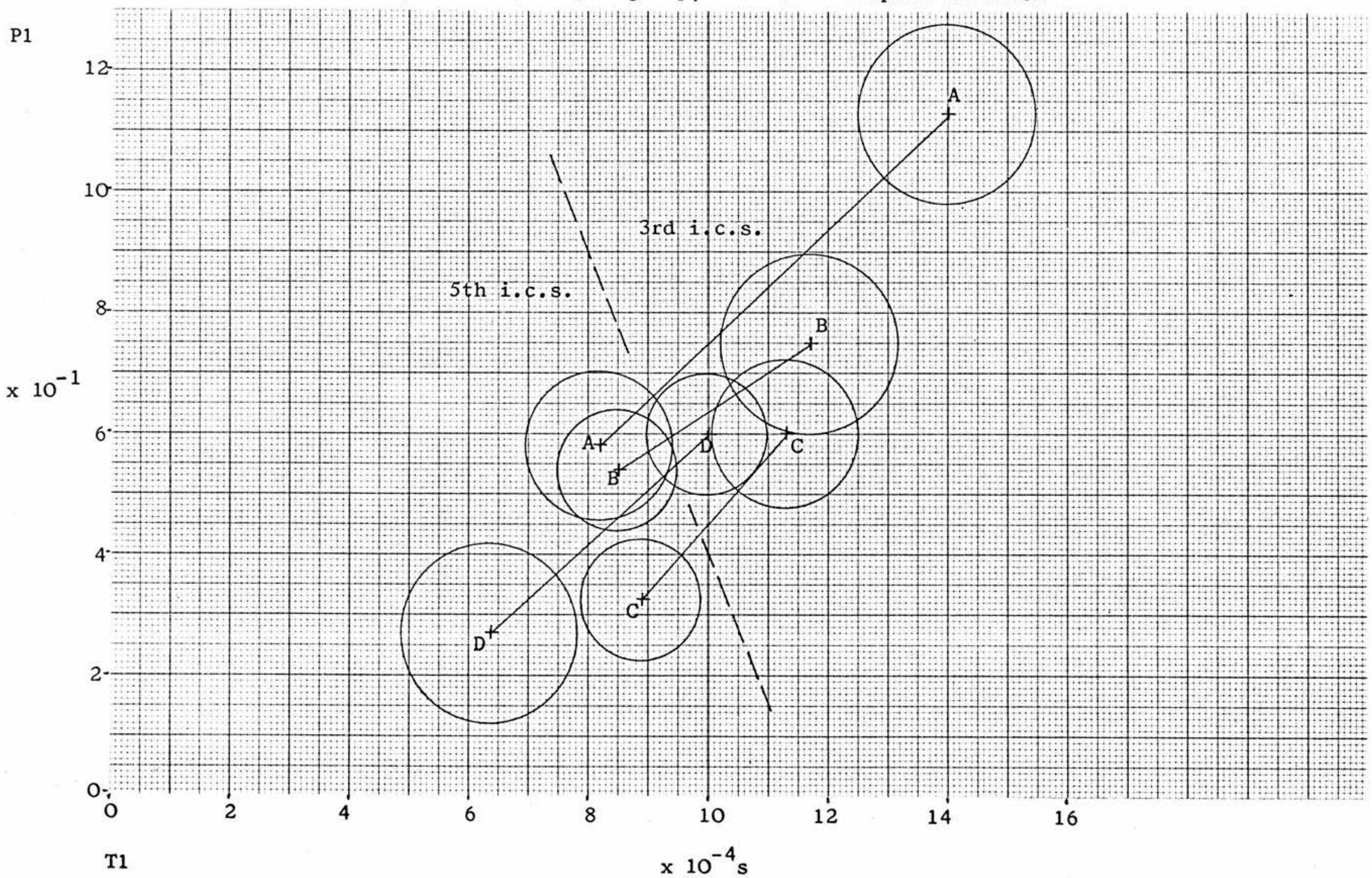
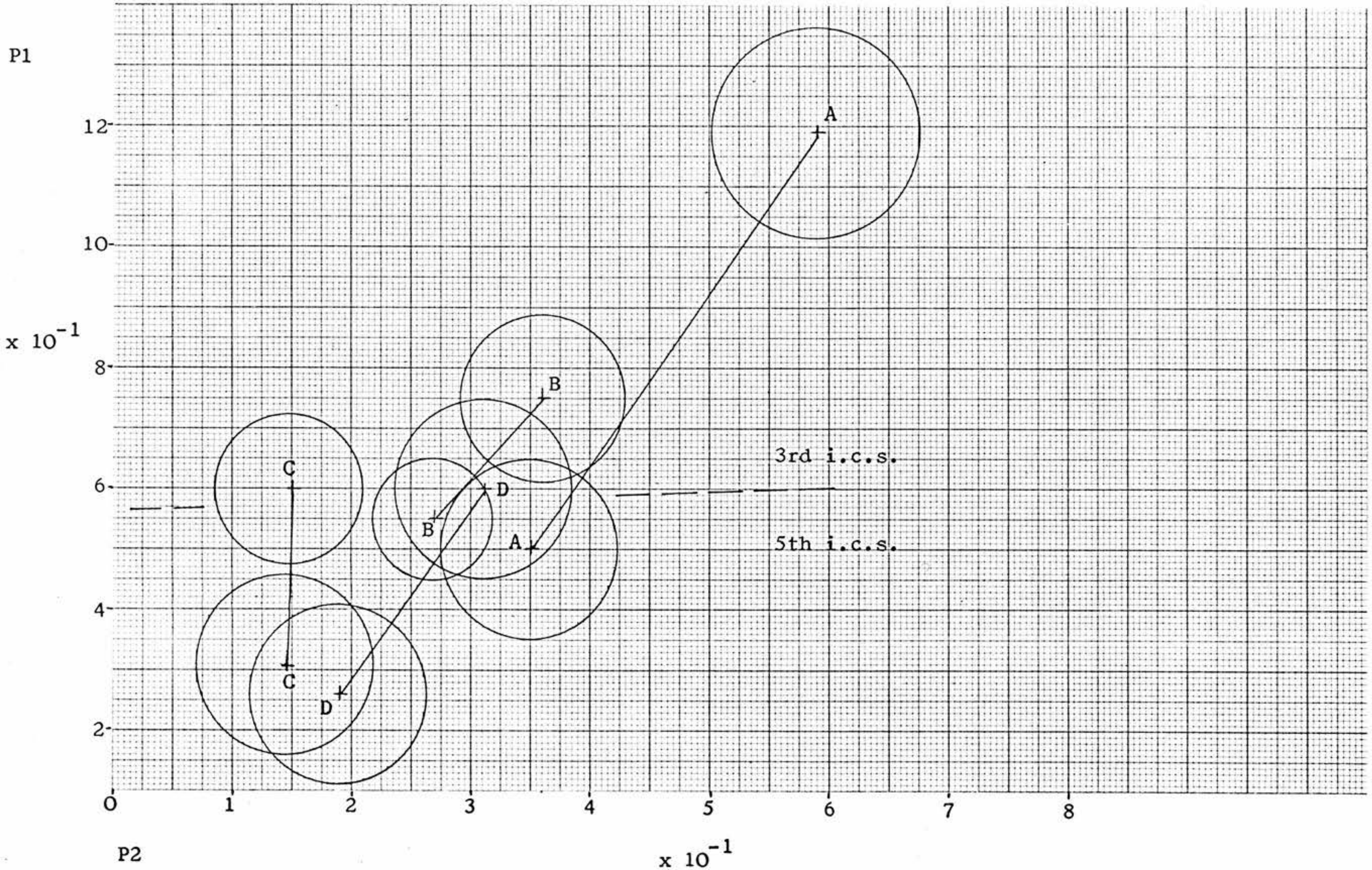


Figure 7.15 P1 - P2 CLUSTER DIAGRAM FOR DISCRIMINATION OF TWO SOUNDS FROM HEALTHY SUBJECTS

Lung volume - FRC; 20 samples per group; > 90% of samples enclosed



roughly within the range of one to ten.

It should be noted that figures chosen allow a two dimensional separation to be measured directly from the cluster diagrams contained in this chapter.

Satisfactory comparisons can often be obtained using two or three parameters. Cluster separation need not always be measured in four parameters. In subsequent sections of this chapter, two or three parameter cluster separation is used if the additional parameter or parameters did not significantly increase the separation.

7.5 NORMAL PERCUSSION SOUNDS

Although those sounds studied so far were recorded from a healthy subject, it must not be imagined that identical results can be reproduced from any healthy subject. The results can only be regarded as typical. Person-to-person variation in percussion sound is large.

To investigate the extent of parameter variation from one healthy subject to another, an analysis was made of the sounds from four such subjects. The sounds analysed were those from the third and fifth intercostal spaces with the subjects' lung volume at FRC.

Results showing the relationship between three parameters (P1, P2 and T1) are shown in figures 7.14-15. The circles enclose more than 90% of sounds in each group.

An attempt has been made to draw a line discriminating between the sounds from the two different positions. This was more satisfactory on the P1-T1 cluster diagram. That it was not a simple matter to discriminate between the two groups highlights the fact that there is a large range of normal percussion sounds. In fact some sounds from one person's third intercostal space were similar to those from

Figure 7.16 LEFT TO RIGHT LUNG PERCUSSION SOUND VARIATION FOR HEALTHY SUBJECT

Lung volume - FRC; Position - 4th i.c.s.; 20 samples per group; > 90% of samples enclosed

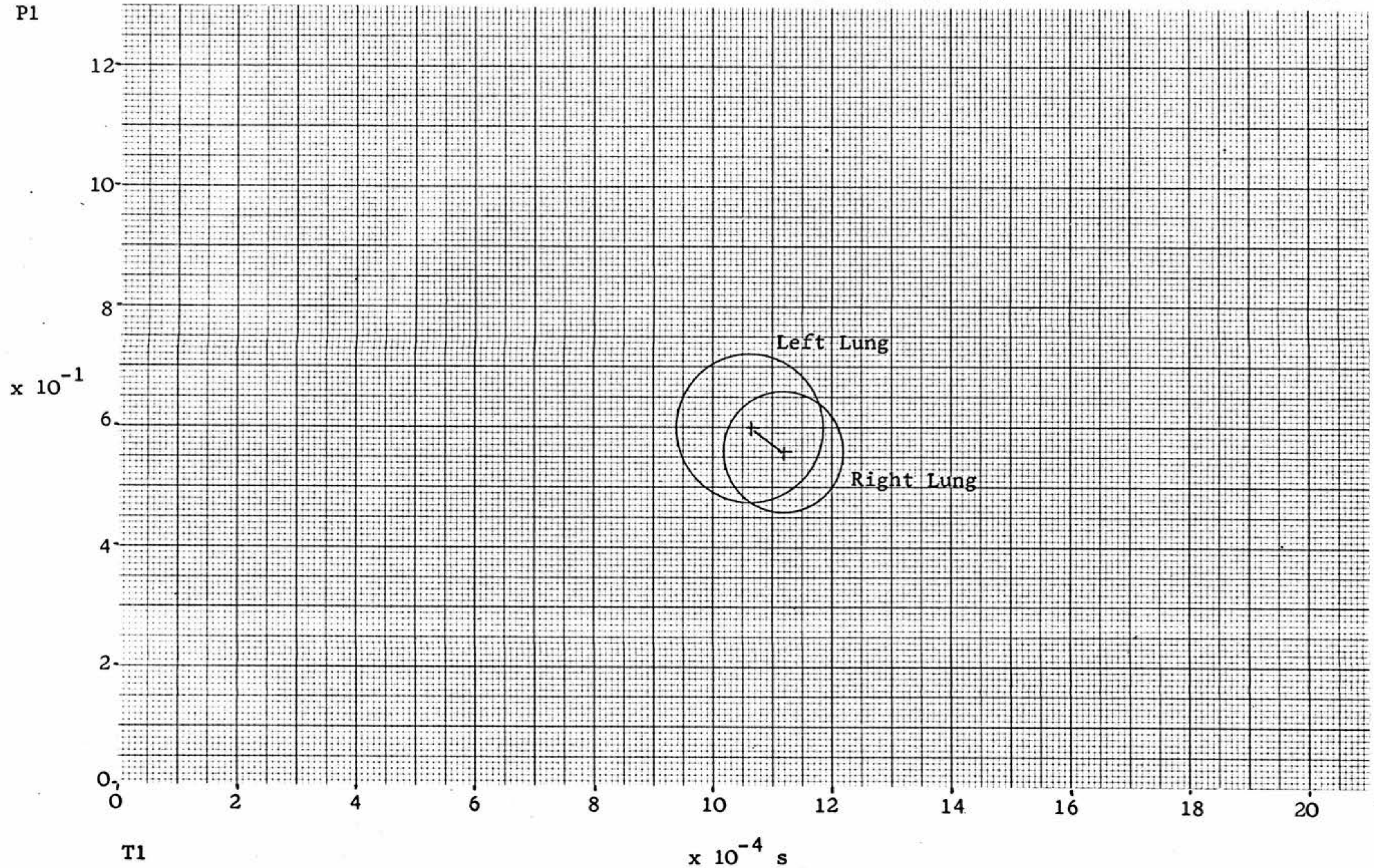
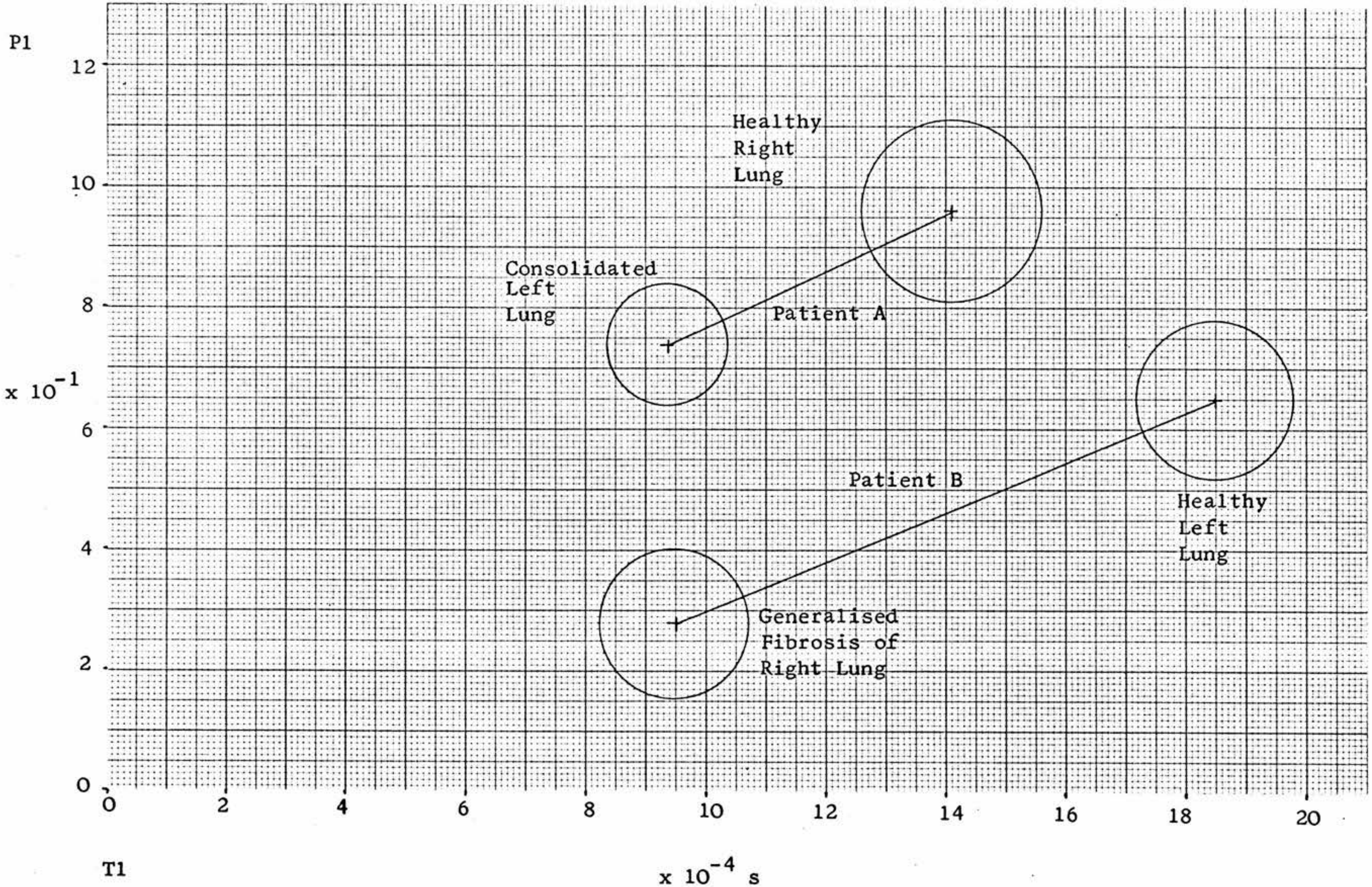


Figure 7.17 DETECTION OF DISEASED LUNG TISSUE

Lung volume - FRC; Position - 4th i.c.s.; 10 samples per group; > 80% of samples enclosed



another's fifth intercostal space.

The magnitude of the changes in sound from one intercostal space to the other were not consistent either. For example, comparing subject A with subject B it can be seen that A has a much larger difference between the two sounds.

The table below illustrates this better by listing quantitatively the change in parameters between those sounds measured over the third and fifth intercostal spaces. The distance of seaparation between clusters is measured in three dimensions (P1, P2, T1).

Subject	Cluster separation in P1, P2, T1 space (sounds from 3rd and 5th i.c.s.)
A	10.2
B	4.2
C	3.8
D	5.1

7.6 DETECTION OF DISEASE

Medical practitioners detect disease by noting abnormal changes in sound quality while percussing over a diseased area. The large person-to-person variation of percussion sounds shows that a comparative technique of this type is the only one possible. An absolute assessment of a single percussion sound reveals very little useful information. This abnormal change in the quality of the sound is listened for while percussing down or across the thorax, or by comparing one lung with the other.

Comparing the percussion sound from right and left lung of a healthy subject produced the results shown in figure 7.16. The sounds were taken from the fourth intercostal space with the lung volume at its FRC.

As can be seen the two cluster groups overlap considerably. The sound from one lung was very similar to that from the other lung. This time only one cluster diagram is included, since P1 and T1 yielded the best discrimination of sounds when diseased tissue was later considered. (A cluster separation of 0.7 units was measured.)

Next, similar measurements were made for two hospitalised patients known to have a significant lesion in one lung. Figure 7.17 shows the cluster diagrams obtained from the results.

Patient A had a healthy right lung and a consolidated left lung. The two clusters of figure 7.17 had separated completely showing that a substantial difference in sound had been measured. The cluster separation this time was 5.2 units.

Patient B had generalised fibrosis of the right lung. The left lung, although healthy, had become slightly 'hyperresonant' due to its extra breathing load. In this latter case the waveform was a few milliseconds longer than the 15 ms previously obtained. The parameters, nevertheless, were still measured as defined. Cluster separation was measured as 9.7 units.

The table below lists the two dimensional separation (P1 and T1) of the left and right lung sound clusters, and allows the results from the patients to be compared with that of the normal subject.

Subject	Cluster separation in P1, T1 space (Sounds from right and left lungs)
Healthy subject	0.7
Patient A	5.2
Patient B	9.7

The difference in sounds between the two lungs for patient A was more than seven times that of the healthy subject. For patient B, it was more than thirteen times. Such encouraging results gave further weight to the possibilities of the four parameter measurements.

CHAPTER 8

SUMMARY AND CONCLUSIONS

Percussion, although it is an extremely useful diagnostic aid, is still poorly understood by the medical profession. A great deal of research effort was at one time expended in the investigation of percussion, but the results obtained have been found to be unsatisfactory.

Therefore an incentive existed for beginning the investigation of percussion afresh.

8.1 PHYSICAL ASPECTS OF PERCUSSION

Much of the work on percussion has been directed at explaining how percussion sounds are produced. Many attempts have been made to explain the sounds on the basis of Helmholtz resonance theory. (Resonance in this case refers to natural modes of vibration.) However, in this present study no such resonance could be observed on the sound pressure waveforms, except for 'tympanic' sounds.

Hence in the explanation of the 'resonant' percussion sound, the waveforms were first studied, and then from the observations, a theory was developed. A consideration of sound propagation led to the expectation that the pressure wave should be directly proportional to the acceleration of various points on the chest. Two principal components were isolated - one due to those ribs with which the pleximeter was in contact and the other to the ribs adjacent to the percussion area.

Therefore, although the air cavity of the lungs was not found to display any Helmholtz resonance, the presence of air in the lungs, through its effect on the mechanical impedance of the thoracic contents as a whole, appears to alter the waveform of the transient acceleration of the ribs produced by the percussion blow, and hence also the resulting sound pressure waveform.

The theory developed to explain the production of the sound proved to be a far more satisfactory theory than that of Helmholtz resonance, which was based on attempts to identify frequency components when listening to the sounds - a subjective method. The present theory, on the other hand, was based on a detailed study of the waveform, which was then correlated with actual measurements of acceleration.

8.2 PERCUSSION SOUND PRESSURE WAVEFORMS

Not only in connection with the physical aspects of percussion are sound pressure waveforms useful. They deserve a study in their own right.

Authors of textbooks on physical diagnosis have for the most part made no attempt either to produce or reproduce sound pressure waveforms when discussing percussion. Those that do, tend to confuse matters by reproducing extremely dubious waveforms. An example of this is the continued use of Selling's recordings which he made in 1907.

A similar position exists with clinical teaching. The sounds are talked about and listened to, but no attempt is made to allow the medical student to visualise their waveforms.

An example of the benefits which can be obtained using visual waveform illustration can be seen in auscultation. Attempts to listen to the heart sounds developed by the normal mechanical events that occur during the cardiac cycle is made considerably easier if phonocardiography is employed at the same time to produce a trace of the sound waveform being listened to. This is particularly so when attempting to hear splitting of the heart sounds.

As with phonocardiography in auscultation, the percussion sound waveforms can be an immense aid to the understanding of percussion; perhaps because it is often useful to have a pattern which our minds can visualise when we think of the different sounds. In addition, having visual waveforms permanently available allows them to be studied at any time; something which is not possible with the sounds themselves, one reason being that soundshave to be reproduced afresh every time a comparison is to be made. During a lecture prior to clinical teaching, or in a textbook, time can be taken with the illustrations to point out differences between sounds. Not that these differences need necessarily be memorised or measured, but so that, to begin with, they can at least be seen to exist.

If this was attempted the medical student would be presented with proof that distinctive contrasts do exist between sounds before he makes any attempt to listen for the differences himself. He would therefore be convinced from the beginning that there is a variation in sound and would then approach percussion without scepticism about its possibilities.

Perhaps it is not only the medical student who needs his interest stimulated and his mind awakened to the possibilities of percussion. Roberts (1966) claims that "careful percussion is rapidly becoming a neglected art". The trained physician may need reminded of the potential of percussion, which would enable him to approach this old diagnostic aid with a fresh outlook.

Because specific features of the sounds can be examined, and because differences can be compared and contrasted, percussion sounds need no longer remain subjective impressions.

It is therefore suggested that waveforms, similar to those shown in figures 4.2 to 4.6, be used in the study of percussion, both in textbooks and in lecture material.

8.3 QUALITATIVE DESCRIPTION AND ANALYSIS

Before moving on to discuss the description of sounds we must consider the terminology in use. Table 1.1 listed the terms employed by various authors and hence underlined the confusion existing within the field at present. This was especially evident over the use of 'dull', sometimes employed to describe no 'resonance', and at other times, decreased 'resonance'. In this thesis the term 'dull' has been reserved exclusively for those sounds exhibiting no 'resonance', as obtained over those areas where no underlying air cavity is present. It is recommended that this use be continued, since impaired or decreased 'resonance' is a relative term, whereas the condition of no 'resonance' can, without doubt, be separated from that of 'resonance' itself.

In addition, care must be taken when using such relative terms. They must not be used in the description of a single sound, but only when comparing sounds.

A case does, however, exist for retaining the term 'hyperresonant'. Sometimes a 'resonant' sound from a patient can be so highly 'resonant' compared with any typical 'resonant' sound, that it can be identified as being of a different quality from a normal 'resonant' percussion sound without any need for comparing it with another sound.

Nevertheless, 'hyperresonance' is just an extension of 'resonance', as it merges naturally with those sounds which exhibit a high degree of 'resonance'. 'Hyperresonance' must still be classed as a 'resonant' sound.

Apart from the distinctive 'tympanic' sound, the only two terms which need be employed are 'resonant' and 'dull'. Impaired or decreased 'resonance' and highly 'resonant' should only be used when comparing one 'resonant' sound with another and so no attempt should be made to employ these terms in an absolute way.

Descriptions of sounds should be kept simple. However they should not be so brief that their meanings cannot be understood, as happens with the use of terms like 'high' or 'low' and 'soft' or 'dull'.

Ambiguity can arise over the use of the single words 'high' and 'low'. 'High' refers to 'dull' sounds and 'low' to 'resonant' sounds. The simple step of adding 'frequency' to both words - giving 'high frequency' and 'low frequency' - would make the terms more easily understood. However that in itself does not describe the sounds particularly well. Two descriptions will be attempted here, one will try to relate the percussion sounds to sounds of everyday experience, and then, because vagueness could still be levelled at such a description, spectrum analysis will be considered to explain frequency content on a more scientific basis.

'Tympanic' sounds are the most musical of percussion sounds. They, unlike the other sounds, have a definite frequency component, and hence musical pitch can be used to define them. Their quality is perhaps nearest to that of a large drum. Dullness, because it is such a short sound sensation is more difficult to define. One of the terms often used to designate extreme 'dullness' with, is 'stony dull'; it is almost like that sound produced when tapping a stone. 'Resonance' lies somewhere in between 'tympany' and 'dullness'. Its quality can be approached by that of a very well damped drum. Care must be taken with the use of these analogies. No attempt can be made to deduce anything about the nature of the sounds from the analogies. Their only use is in giving the reader an impression of the sounds heard.

Hence a fuller, although perhaps not so medically useful, explanation must be given. The Fourier spectrum of each sound will be considered, as this allows their spreads of energy throughout the sound spectrum to be compared. 'Tympanic' sounds have distinctive peaks in their spectra at those frequencies which define the pitch of the sounds. 'Dullness', because it is a very short sound spreads its energy more than the other sounds. Why it is described as a high frequency sound can only be discovered after considering 'resonance'. 'Resonance' has a distinctive concentration of energy at low frequencies. This explains why it is described as a low frequency sound and also why 'dullness', when compared with 'resonance', is described as a high frequency sound.

Describing sound intensity with 'soft' and 'loud' also leads to confusion, because the resulting intensity depends to a great extent on the force of the percussion blow. A better way of qualitatively describing this feature is obtained by assessing the ease with which the sound can be generated. This takes account of both force and intensity. The 'resonant' sound is relatively easily produced because the percussion blow can easily set the compliant air-cushioned ribs in motion. No such air cavity underlies the ribs over a 'dull' area and so the sound becomes harder to generate. Such a description helps relate the sound heard to the tactile sensation.

From the observation made on the tactile sensation it is suggested that the now prevalent description be abandoned. No justification could be found for the continued use of 'sense of vibration' as the descriptive term for the tactile sensation. Instead, it is recommended that the original description - 'sense of resistance' - be returned to. Although it could not be claimed that 'vibrations' were being detected, the sensitivity of the finger must not be belittled in any way. Through its sense of touch the finger was able to discriminate between sounds which have been shown to be less than 15 ms long.

During the percussion blow the pleximeter finger was found to be assessing the compliance of the chest at the point being percussed. It does not, as is presently thought, detect the frequency of thoracic vibrations.

'Dull' sounds accompany a high hard resistance, while those of 'resonance' are associated with a firm but springy resistance.

Another important aspect of the description of percussion sounds came from the study and comparison of the various waveshapes. This was achieved both by a visual inspection of the waveshapes and by listening tests to find out which features of the waveforms the ear considered to be important for recognising the sounds and for distinguishing them from others.

The remarkable capabilities of the human ear were demonstrated by the listening tests. Changes in the sound pressure waveforms, when the waveforms were less than 15 ms long, could be easily detected.

Clearly some finite part of the energy of the resonant sound is contributed more than 15 ms after the blow. Room reflections have frustrated attempts to examine this late 'tail' of the 'resonant' sound despite the use of averaging techniques to remove noise. Though the use of an anechoic chamber would allow further study of these late components, they contribute little to the characteristic sound of the 'resonant' waveform since it has been shown that truncating the waveform at 15 ms leaves the sound easily recognisable for what it is.

In concluding this section on the description of percussion sounds it would be useful to take each of the important sounds in turn and summarise their most basic features. It should be remembered that the sounds are qualitatively described relative to each other.

'Tympany': A sound which is nearer to having a distinctive musical quality than any of the others. It can be acceptably described by a single frequency component. Frequencies from 200 Hz to 600 Hz have been detected in the healthy subject. Its waveform consists of a large initial rarefaction pressure spike followed by a waveshape similar to an exponentially damped sinusoid. The damping is always high - average time constant of about 5 ms. Of all the percussion sounds this one has the longest duration. It could still be detected above sound reflections after 40 ms. The sound is extremely easy to generate.

Table 8.1 DESCRIPTION OF PERCUSSION SOUNDS

	'Tympany'	'Resonance'	'Dullness'
Waveform Description	Primarily a damped single frequency component	Rarefaction pressure spike and two compression peaks	Rarefaction pressure spike only
Duration	Can still be detected after 40 ms	Approximately 15 ms	Less than 3 ms
Frequency Content	Dominant single frequency	Assessed by ear to have a low frequency component compared with 'dullness'	Assessed by ear to be high in frequency compared with 'resonance'
Comparative Ease of Generation	Very easy	Easy	Hard
Tactile Sensation	Very low resistance	Firm springy resistance	High hard resistance

'Resonance': A sound which is interpreted by the ear as being a low frequency sound. Its waveform consists of an initial rarefaction pressure spike followed by two compression peaks. Its waveform is considerably smoother than that of the 'dull' sound. The useful energy of the sound is contained within 15 ms. The sound is easy to generate. By tactile sensation, a firm but springy resistance is felt.

'Dullness': A sound which is interpreted by the ear as having a high frequency characteristic. Primarily its waveform consists of a rarefaction spike and very little else. The useful energy is contained in less than 3 ms. It is not so easy to generate, and because of the high percussion force required, a substantial amount of finger resonance is generated. By tactile sensation a hard resistance is felt.

Table 8.1 summarises these descriptions, and enables the sounds to be compared.

In spite of the present confusion in medical textbooks, percussion sounds can readily be illustrated and explained in simple easily understood vocabulary.

8.4 QUANTITATIVE ANALYSIS

Some physical phenomena lend themselves to a simple one parameter measurement. Temperature is one example. However, most phenomena defy simple analysis since they require the measurement of more than one parameter.

The first step in an analysis of this type must always be to decide which features are to be measured. For example, consider the analysis of colour. A colour has three characteristics; brightness, hue and saturation. Brightness is that characteristic which is reproduced in black and white photography. Together the other two characteristics give the chromaticity of the colour. Hue comes from our awareness of the differences in light wavelength. Saturation is a measure of the difference between a colour and white. How then were colours to be quantitatively analysed so that information contained in these three characteristics was not lost? This was solved by analysing colours into the quantities of three different primary colours present, these colours being defined before analysis was carried out. Upon recombination of the correct quantities of these three primary colours the original colour is reproduced in all its characteristics; in brightness, hue and saturation. Hence the description of colour must be made in three dimensional space, whereas that of temperature only required a single dimension.

The description of sounds likewise poses problems. Much work has been put into the analysis of, for example, the human voice and musical sounds and the quality of musical instruments. Among those features isolated for the analysis of musical sounds are pitch and harmonic content. For musical instruments, wind and fricative noises must also be studied. This time the description must be made in more than three dimensions, i.e. in hyperspace.

Similarly, percussion sounds require a hyperspace description. The important primary features of the sounds were isolated using a series of listening tests. Then those features had to be described in a way which allowed them to be measured. Unlike the previously considered descriptions of pitch and harmonic structure which are carried out in the frequency domain, the analysis of percussion sounds was achieved in the time domain; the percussion sounds being so short that their distinctive features could be more easily observed in the time domain.

A four parameter model which allowed the sounds to be quantitatively analysed was produced. Proof of any model lies in its ability to reproduce the original. For instance, in the analysis of colour, the measurements made had to enable the original colour to be reproduced. Similarly, the validity of the model for the analysis of percussion sounds had to enable them to be reproduced. An electronic simulator designed to reproduce percussion waveforms from the four parameters already measured was used to prove both visually and acoustically the validity of the model.

Measurement of the four parameters of a large number of sounds proved that there are measurable differences in the sounds. In fact the values of the parameters were found to change quite markedly with such variables as the volume of air in the lungs or with position of percussion on the chest.

No claim is made that the parameters selected are the only possible ones. However, since their use enabled different sounds to be measured it can be claimed that important features were being analysed and that a useful four parameter model had been obtained.

The measurements showed that there is no single percussion sound which characterises a normal healthy person. Indeed, the variation in sound from one healthy subject to another was considerable. Hence the medical practitioner's method of examining a patient by noting the variation in sound with position on the chest is the only possible method.

A final evaluation of this quantitative measurement technique was obtained with the satisfactory detection of diseased tissue.

Quantitative analysis however requires instrumentation. Some diagnostic aids lend themselves to instrumentation. Consider electrocardiography, in which electrodes must be strapped to the patient and a monitoring device made available. In such a case it is only natural that electronic analysis of e.c.g.s be attempted.

Percussion provides no such incentive. Since it is so easy to use, requiring no tools other than the fingers and a trained ear, there is insufficient justification for encumbering the physician with any type of equipment. This applies equally to the hospital physician on his ward round and to the general practitioner in his surgery or on home visitation.

Hence it is not the intention of this research project to propose that percussion should be automated or mechanised in any way, even in the analysis. (At least not for routine work.)

It is hoped that the outcome of the project will have a far greater value than that of introducing an analysis technique which only a few will attempt. Instead, some benefit can be reaped by all physicians and medical students.

Since percussion can be more easily understood, its sounds simply described and the physical aspect of it more fully known, the medical profession can be better prepared to extract the maximum value from such a simple aid, especially since quantitative analysis has provided an added incentive by proving that small changes in percussion sounds can not only be detected, but also measured.

APPENDIX A1

RELATION BETWEEN SOUND PRESSURE AND MOTION OF SOURCE

A1.1 SYMBOLS

p	acoustic pressure or excess pressure at any point.
u	particle velocity.
ρ_o	constant equilibrium density of medium.
c	velocity of propagation of wave.
∇^2	Laplacian operator.
f, f_1, f_2	arbitrary functions.
a	radius of sphere.
λ	wavelength.

A1.2 THE WAVE EQUATION

Sound pressure waves have been shown (see Kinsler and Frey, 1962) to obey the general wave equation

$$\frac{\delta^2 p}{\delta t^2} = c^2 \nabla^2 p \qquad \text{A1 - 1}$$

where the pressure is related to the particle velocity by

$$\text{grad } p = -\rho_o \frac{\delta u}{\delta t} \qquad \text{A1 - 2}$$

It should be noted that the particle velocity of any point at the surface of a source equals the velocity of that point on the source.

A1.3 SOLUTION OF WAVE EQUATION

A1.3.1 Plane Waves

In Cartesian coordinates the Laplacian takes the form

$$\nabla^2 = \frac{\delta^2}{\delta x^2} + \frac{\delta^2}{\delta y^2} + \frac{\delta^2}{\delta z^2}$$

Hence the Laplacian and wave equation for a wave travelling in the x direction become

$$\nabla^2 p = \frac{\delta^2 p}{\delta x^2}$$

$$\frac{\delta^2 p}{\delta t^2} = c^2 \frac{\delta^2 p}{\delta x^2}$$

The wave equation is satisfied by

$$p = f_1 (ct - x) + f_2 (ct + x)$$

Considering only the wave which travels in the positive x direction

$$p = f (ct - x) \qquad \text{A1 - 3}$$

$$\frac{\delta p}{\delta x} = - f'$$

From equation A1 - 2

$$f' = \rho_o \frac{\delta \mu}{\delta t}$$

$$f' \, dt = \rho_o \left(\frac{\delta \mu}{\delta t}\right) dt$$

$$\frac{f}{c} = \rho_o \, u$$

$$f = \rho_o \, c \, u$$

$$p = \rho_o \, c \, u \qquad \text{A1 - 4}$$

A1.3.2 <u>Spherical Waves</u>

In spherical coordinates

$$\nabla^2 = \frac{\delta^2}{\delta r^2} + \frac{2}{r} \frac{\delta}{\delta r} + \frac{1}{r^2 \sin \Theta} \frac{\delta}{\delta \Theta} (\sin \Theta . \frac{\delta}{\delta \Theta}) + \frac{1}{r^2 \sin^2 \Theta} \frac{\delta^2}{\delta \psi^2}$$

Since p is a function of r only

$$\nabla^2 p = \frac{\delta^2 p}{\delta r^2} + \frac{2}{r} \frac{\delta p}{\delta r}$$

$$\frac{\delta^2 p}{\delta t^2} = c^2 \left(\frac{\delta^2 p}{\delta r^2} + \frac{2}{r} \frac{\delta p}{\delta r} \right)$$

$$= c^2 \left(\frac{1}{r} . \frac{\delta^2 (rp)}{\delta r^2} \right)$$

which satisfies

$$rp = f_1\,(ct - r) + f_2\,(ct + r)$$

$$p = \frac{1}{r} \cdot f_1\,(ct - r) + \frac{1}{r} \cdot f_2\,(ct + r)$$

Considering the diverging wave

$$p = \frac{1}{r} \cdot f\,(ct - r) \qquad \text{A1 - 5}$$

$$\frac{\delta p}{\delta r} = -\frac{1}{r^2}\,f - \frac{1}{r}\,f'$$

From equation A1 - 2

$$\frac{1}{r^2}\,f + \frac{1}{r}\,f' = \rho_o \frac{\delta u}{\delta t} \qquad \text{A1 - 6}$$

From A1 - 5

$$f = r\,p$$

and $\dfrac{\delta p}{\delta t} = \dfrac{1}{r} \cdot f'\,c$

$$f' = \frac{r}{c} \quad \frac{\delta p}{\delta t}$$

Substituting into A1 - 6 gives

$$\frac{p}{r} + \frac{1}{c} \quad \frac{\delta p}{\delta t} = \rho_o \frac{\delta u}{\delta t}$$

$$cp + r\,\frac{\delta p}{\delta t} = \rho_o\, c\, r\,\frac{\delta u}{\delta t} \qquad \text{A1 - 7}$$

A1.3.3 <u>Pressure at Surface of Simple Source</u>

If the vibrating sphere is very small, having its radius small compared with the wavelength of the sound radiated, the sphere is called a simple source of sound.

At the surface of the simple source

$$r = a$$

and from equation A1 - 7

$$cp_a + a\frac{\delta p_a}{\delta t} = \rho_o\, ca\, \frac{\delta u_a}{\delta t}$$

Considering $a\frac{\delta p_a}{\delta t}$ for sinusoidal waves

$$a\frac{\delta p_a}{\delta t} = j\omega\, a\, p_a$$

$$= j\, 2\pi \frac{c}{\lambda}\, a\, p_a$$

$$\therefore\ c\, p_a\left(1 + j\, 2\pi\frac{a}{\lambda}\right) = \rho_o\, c\, a\, \frac{\delta u_a}{\delta t}$$

but a λ, and so returning to the general case

$$\therefore\ c\, p_a = \rho_o\, c\, a\, \frac{\delta u_a}{\delta t}$$

$$p_a = \rho_o\, a\, \frac{\delta u_a}{\delta t} \qquad \text{A1 - 8}$$

It can be seen that the sound pressure resulting from the vibration of a simple source is directly proportional to the acceleration of that source.

When this pressure wave is propagated it obeys the spherical wave equation (A1 - 5), and so attenuation and a time delay are the only changes introduced. Hence if the time delay is taken into account, the pressure at some point in the path of the propagated wave is still directly proportional to the acceleration of the source.

Summarising the main equations:

Plane wave	$p = \rho_o\, c\, u$
Spherical wave	$cp + r\,\frac{\delta p}{\delta t} = \rho_o\, cr\,\frac{\delta u}{\delta t}$
Simple source	$p_a = \rho_o\, a\,\frac{\delta u_a}{\delta t}$

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